1: Executive summary

Our objective is to design and fabricate a vacuum tube stereo power amplifier where the audio path is fully analog and constructed with operational amplifiers in the signal processing section, yet uses digital potentiometers for volume and control of an op-amp based graphic equalizer. After processing, the audio signal is passed to a vacuum tube differential amplifier that also provides most of the signal gain. The 180-degree out of phase signal pair drives opposing halves of a vacuum tube push-pull amplifier to provide the power gain. An impedance transformer matches the very high output impedance of the amplifier with the 8-ohm loads of common speakers

2: Project Description

A vacuum tube audio amplifier for which the entire signal path is analog but the audio parameters are digitally controlled via a touch screen graphical user interface which also displays visualizations of the amplitude, frequency and phase characteristics of the audio signals.

2.1: Motivation

This project has several motivations: the first motivation is the desire to work successfully as an age-diverse team. Our team is unique this semester in that one of the team members, Stephen, already is an electrical engineer by trade and has nearly a lifetime of experience in the electronics industry, over 32 years in fact. And, at age of 53, Stephen is considerably older than everyone in class. It is our desire as a group to share a unique design experience. For Jason and Rafael, the hope is to benefit from the experience and insight that an older team member provides. For Stephen, the hope is to be able to practice leadership (of Stephen's engineering years, less than one year total can be said to represent management) as well as the insight of seeing through younger eyes again for the fresh perspective to problem solving they can provide. The synergy of our team will make this project a very exciting one.

The second motivation is nostalgia: it is likely that most people under 40 have never seen a piece of electronic hardware designed around vacuum tubes, let alone seen or held a vacuum tube in their hands. When Stephen was born in 1960, virtually all electronic circuitry was vacuum-tube based. It was not until the late 1970's that TV sets began to be dominated by solid-state circuitry, and Stephen worked on a fair share of them. But around 1980, solid state circuitry began to dominate. Electronics made with vacuum tubes pretty much died out around that time.

For virtually all engineers if not all of society in general, a vacuum tube embodies the very idea of obsolescence. *Who* designs anything with vacuum tubes anymore? Who would even *want* anything made from tubes? Both of these are

fair questions, but as well see in this project, vacuum tubes fulfill a very small niche need that cannot be satisfied by anything that chops the sound into little pieces no matter how fine. In fact, the answer to both questions is rather surprising: *Lots of people*. They just represent a very small (but not zero-sized) segment of society. Indeed, the reader will be amazed at the amount of money that someone will spend on equipment made of vacuum tubes, and utterly astounded at the "lost" advantages that analog processing can provide.

Our third motivation comes from Stephen who prefers the sound of pure analog over digital (and longs for the chance to rebuild one of the older "cathedral" radios someday); who has always wanted to build a vacuum-tube based project, and who wanted to design a modern microcontroller based project for years, but lacked a suitable candidate. This project will accomplish all of these goals, and provide invaluable project experience in analog and digital design for all team members. The result will be a 100% analog power amplifier, including vacuum tubes in the power output stages, yet is fully digitally controlled.

While many people will find the idea of using vacuum tubes quaint, they are still favored by many musicians and high-end audio enthusiasts for their mellower sound and low-distortion characteristics. This effect, known as "tube sound", is difficult to quantify and seems to be largely anecdotal with professors and detractors on both sides of the issue. No authoritative data seems to be readily available, but the preponderance of opinion is that the desired sonic effect comes from the "soft clipping" characteristics of vacuum tube amplifiers which emphasize even-order harmonics, as opposed to solid-state designs that tend to produce odd-order harmonics when they clip during musical peaks. (It should be noted that even-order harmonics are simply the same musical note at a higher octave, while odd-order harmonics of any given musical note are a different, generally non-standard, musical note – See Table 1 for a few examples).

Frequency	Harmonic Number	Musical Note
440 Hz	1 (fundamental)	A in 4th octave
880 Hz	2nd	A in 5th octave
1320 Hz	3rd	Approx. E in 6th octave
1760 Hz	4th	A in 6th octave
2200 Hz	5th	Approx. C in 7th octave

Table 1

2.2: Design goals

The goal is to design, document, and fabricate an all-analog vacuum-tube pre-amplifier and power amplifier, yet be all digitally controlled. The finished product must incorporate a modern touch screen / LCD-based Graphical User Interface (GUI) control for all functions except power. The LCD should produce an entertaining display that reacts to the audio signal in a similar manner to Microsoft Windows Media Center when not being used for control.

The envisioned end use of the finished product is a residential setting, such as a living room or family of a typically-sized suburban home. It is anticipated that the product will be placed on an entertainment center shelf, a bookcase, or perhaps a piece of furniture such as an end table. See Figure 1 for a typical setting. The product is intended for serious and critical music listening, an activity that the end user would be doing as a primary activity. The volume produced should be “room filling” but not so loud as to hinder conversation. Based on experience with previous commercially-made stereo receivers, the team agreed that 10 watts per channel minimum, with a goal of 15 watts per channel would suffice.

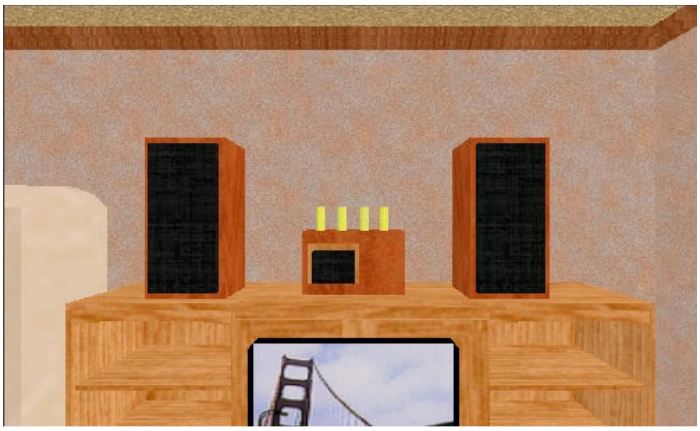


Figure 1: The intended use of the finish product, shown with typical stereo speakers (Created by Joshua Nichols – son of one of the authors).

While a vacuum-tube-based amplifier is desired for its clipping characteristics, it should not introduce excessive distortion when not clipping. This means that for much of the signal, the distortion should be minimal – comparable if not better than a solid state audio amplifier. The amplifier should contribute as low noise as possible (beyond what is already present in the signal) and introduce absolutely no 60 or 120 Hz components, commonly known as “hum”. The audio signal path must be completely analog. Ideally the entire audio path could be processed using vacuum tubes, as will be seen in the commercial designs presented later in this document, but that would greatly complicate digital control due to the need to interface the high-voltage tube circuits to the low-voltage microprocessor circuit. Solid state operational amplifiers will be used as the basis for a graphic equalizer.

The finished product will not have any physical controls, except for a toggle-type power switch. Due to the use of AC-power-line derived voltages of as high as 500 volts DC, safety is paramount. Accordingly, the finished product shall be free of electrical hazards. All exposed metal surfaces shall be grounded, and the AC power input shall be fused. The physical power switch shall break the hot and neutral power signals when in the “OFF” position. Due to the exposed delicate vacuum tubes that could expose high voltages if broken, the finished product is not intended to be used by small children.

2.3: Project Requirements and Specifications

Within the project there are multiple hardware and software components which need to interact and therefore must meet their requirements and specifications.

2.3.1: Hardware

The performance requirements are summarized in Table 2.

Requirement	Value	Condition
Number of audio channels	2 (stereo)	
Output power rating	10 Watts Root Mean Square (RMS)	Per channel at 1000 Hz without clipping
Input impedance	10 K ohms	Per channel
Output impedance	8 ohms	Per channel
Bandwidth	20 Hz to 20 KHz flat ±3dB	As measured at a moderate output level relative to the input signal level
Total Harmonic Distortion, low signal level	0.5%	When measured at a number of frequencies 100 Hz to 5 KHz 12dB below maximum output
Total Harmonic Distortion, high signal level	2.5%	When measured at the onset of clipping at a selection of audio midrange frequencies
Hum	None detectable	
No ungrounded exposed metal surfaces	Full compliance	Measure continuity from each surface to ground

Table 2: Performance requirements of the finished product

The finished project shall be housed in a finished cabinet of contemporary design suitably designed to withstand normal wear and tear typically encountered during residential use. The cabinet shall use stained wood for the front and side surfaces, and sheet aluminum for the top, back and bottom surfaces. As a goal, the completed project shall weigh a maximum of 25 pounds. See Figure 2 for an early concept image of what the completed project might look like. Like all contemporary commercially made vacuum tube amplifiers, the vacuum tubes shall be mounted such that they protrude from the top surface. Mounted this way the glowing tubes will lend a striking appearance to the project, as well as help with the heat dissipation, as doing so will keep the over one hundred watts of heat generated by the tubes on the outside of the project.

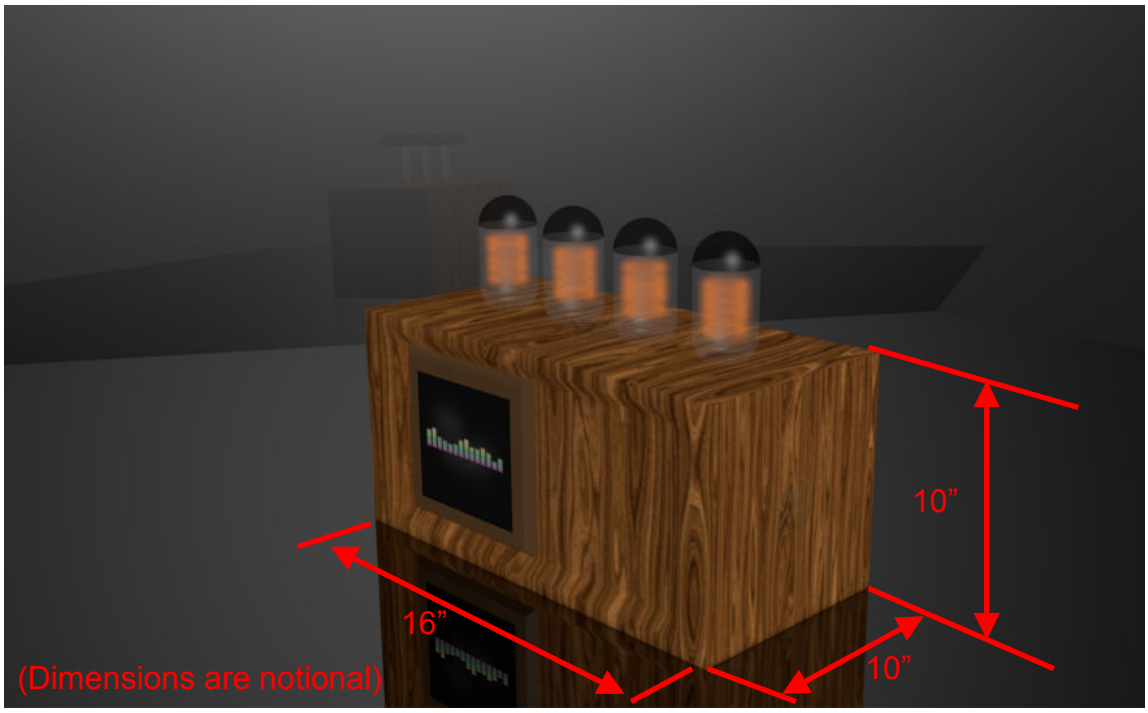


Figure 2: Early concept of the finished product. (Created by the authors)

The design in Figure 2 proved to be impractical to build during the limited time available in Senior Design 2. A simpler box was fabricated for the purpose of presentation and appears in Figure 3. The tubes are mounted on the Audio Processor CCA, with vent holes cut in the top surface directly above them.

Figure 3: The as-built finished product

The vacuum tubes are inherently delicate and vulnerable; hence the finished product is not intended for frequent moves. The six vacuum tubes will be socket-mounted to enable them to be removed and packed for transport or long-term storage. The power transformer(s) and impedance-matching transformers will be mounted directly to the lowest horizontal surface of the chassis, and make connection via twisted shielded wire. The transformers will be encased in a metallic sub-enclosure ideally made of a high-permeability metal such as steel, Permalloy 80 or mu-metal (in order of increasing permeability). The finished project shall incorporate some means to measure the internal chassis temperature and provide a warning on the display if a preset limit temperature is exceeded.

The entire audio path through the project shall be analog. The graphic equalizer circuits shall use high-performance linear operational amplifier devices. A six-band graphic equalizer is preferred. The output of the audio processor shall be a minimum of 5 volts peak without clipping. The power amplifier shall use vacuum tubes for the amplification devices and require no more than 5 volts peak to provide the full unclipped rated output power. All audio parameters shall be digitally controlled: Input source select switching, volume, balance, etc. using a touch screen operated GUI. Any 8, 16 or 32-bit microcontroller may be used for the operating software provided sufficient serial port, parallel port and A/D conversion ports are available.

The high-voltages required for the vacuum tube will generated from a separate independently-mounted supply. Assuming 30% power efficiency for the push-pull amplifier stage, which will use the bulk of the power, 200 watts of 110 volt AC power is required. Input power shall be provided via an IEC-320 C13 connector (a standard power cord of the type used by computers shall connect from the AC socket to input power connector.) The ground wire shall terminate at the metal chassis and be connected to all user-accessible metallic parts of the finished project. A toggle-type power switch shall be provided that completely isolates the AC power line from the internal circuits when placed in the “OFF” position. The toggle switch shall be mounted on the front panel, the same as the touch screen. A 1¼” x 1¼” cylindrical fuse shall be provided in a suitable holder between the switch’s “load” terminals and the power supply input terminals, on the back panel (opposite the touch screen).

Most non-commercial vacuum tube amplifiers are built directly on a metal chassis with the vacuum tubes and all of the parts more or less mounted individually. In such a construction, all of the wiring is point-to-point, and is prone to noise, oscillation, and 60 Hz hum pickup. To combat such maladies, this project will be designed and built with due attention and strict control of grounding, shielding, and isolation: The digital and analog parts of the circuit will be separately grounded, with just a single ground connection between them. The analog circuits will be fabricated on a printed circuit board and employ a single-point ground methodology implemented with a very heavy ground bus occupying one

edge of the circuit card assembly (CCA). The vacuum-tubes will be mated to ceramic sockets mounted on directly on the CCA.

The digital circuits will be isolated from the analog circuits by being on a separate printed circuit card in shielded sub-enclosure. Finally, the analog and digital circuits will employ separate voltage regulation. The power supply may be built on the same CCA as the microcontroller, as long as the two circuits are located on opposite sides. All signals passing on or off of the circuit card assemblies shall do so via dual-row rectangular connectors with 0.1-inch. Signals shall pass between the digital and analog circuits by means of optical couplers for digital signals and shielded wire for analog signals. Ribbon cable shall be used wherever feasible.

2.3.2: Software

Upon power up, the finished project shall execute a reset function that performs the following tasks.

1. Displays a splash screen that shows the project name, authors, software revision and date.
2. Sets all graphic equalizers to their center (±0dB) positions. Sets the volume to a low level.
3. Enter the main program loop.

See Figure 3 for an example splash screen.

Within the main program loop, the software will provide an entertaining graphical display that functions when the touch screen is not being touched. Suitable software shall generate this display, providing from three to six “visualizations” that react to the frequency, amplitude and phase characteristics of the sound. The visualizations should last several minutes each and cycle from one to the next automatically. The following are examples of visualizations that could be provided, and not intended to limit the creativity of the designers:

1. Frequency bar-graph display: Shows the real-time and recent-peak amplitude of the input signal.
2. Lissajous pattern: The vertical and horizontal amplitudes and phases are derived from the left and right signal instantaneous signal amplitude and the color derived from the instantaneous dominant frequency.
3. Fourier Display: A 3D-like display showing frequency in the X-axis, the time in the Y-axis and a time averaged amplitude in the Z-axis. In the Y-axis, the display will provide the current spectrum in negative direction (the background) brightly lit and solid, while previous spectrums will appear to come toward the viewer in the positive direction (the foreground) while getting progressively fainter and /or more transparent until disappearing entirely.

4. Running VU meter: Vertical bars that indicate the highest amplitude detected during a (50-millisecond - notional) period that scroll to the right and are displayed in random colors.

See Figure 4 for an illustration of these examples, and the splash screen. Final selection of the visualizations will be made based upon programming feasibility and time limitations.

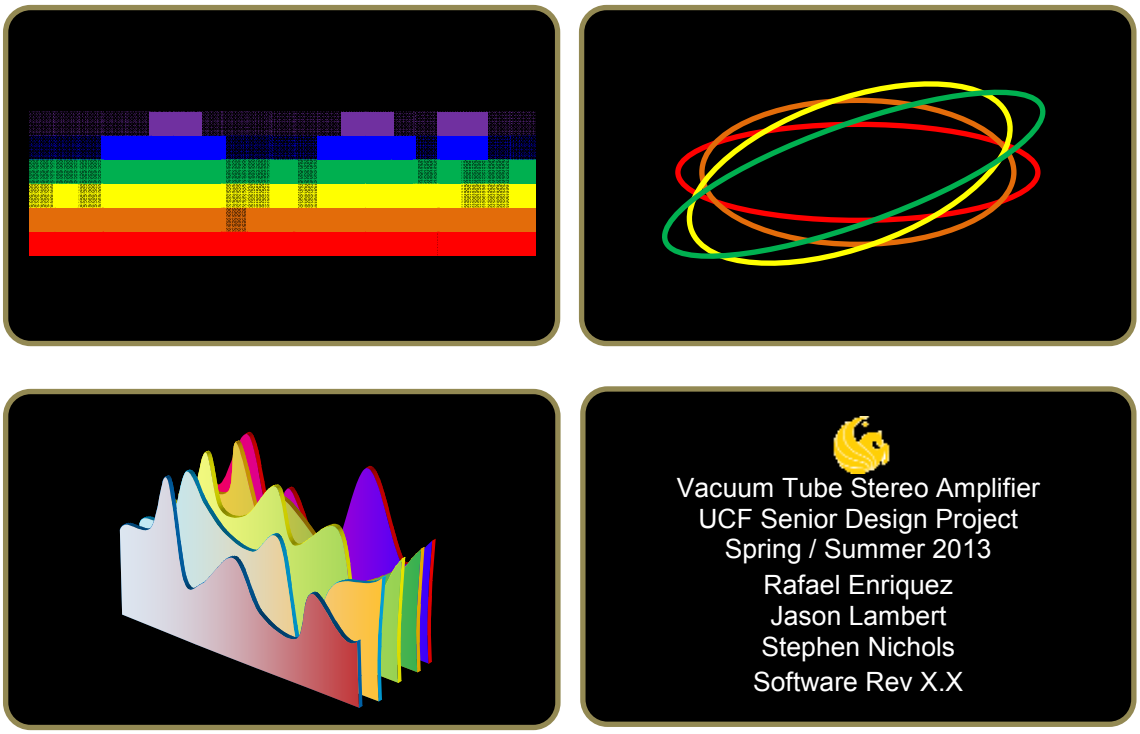


Figure 4: Typical visualizations displayed during the main loop, and the reset screen. (Created by the authors)

While in the main loop, the software shall monitor the touch screen and enter the control mode upon any touch longer than 250 milliseconds duration. The control mode screen shall containing appropriate graphics to serve as virtual controls for input source select, graphic equalizer, volume, balance, presets that are activated by the overlying touch screen. The internal chassis temperature in degrees Fahrenheit shall be displayed in one corner. A warning should flash if the temperature exceeds the set threshold. See Figure 5 for an example control mode screen. The actual layout, graphics, fonts and colors are left to the discretion and imagination of the designers. Different backgrounds may be displayed upon entry to the control mode if desired as long as the layout of the controls remains constant.

Upon touching any control except for settings or one of the preset functions (not limited to four), the software shall update the appropriate hardware registers of the microcontroller to implement the desired control function. Upon touching any

of the PRESET buttons, the software shall adjust all functions according to preset values. Whenever the following sequence of controls occur, SETUP – one of preset buttons – SETEP, the software shall store the current equalizer, volume and balance settings in memory. In between the two SETUP presses, an indication of setup mode shall be provided, and an additional control shall be presented to allow the warning temperature set point to be adjusted. If the second SETUP press does not occur within TBD seconds, the software shall return to control mode without storing any settings.

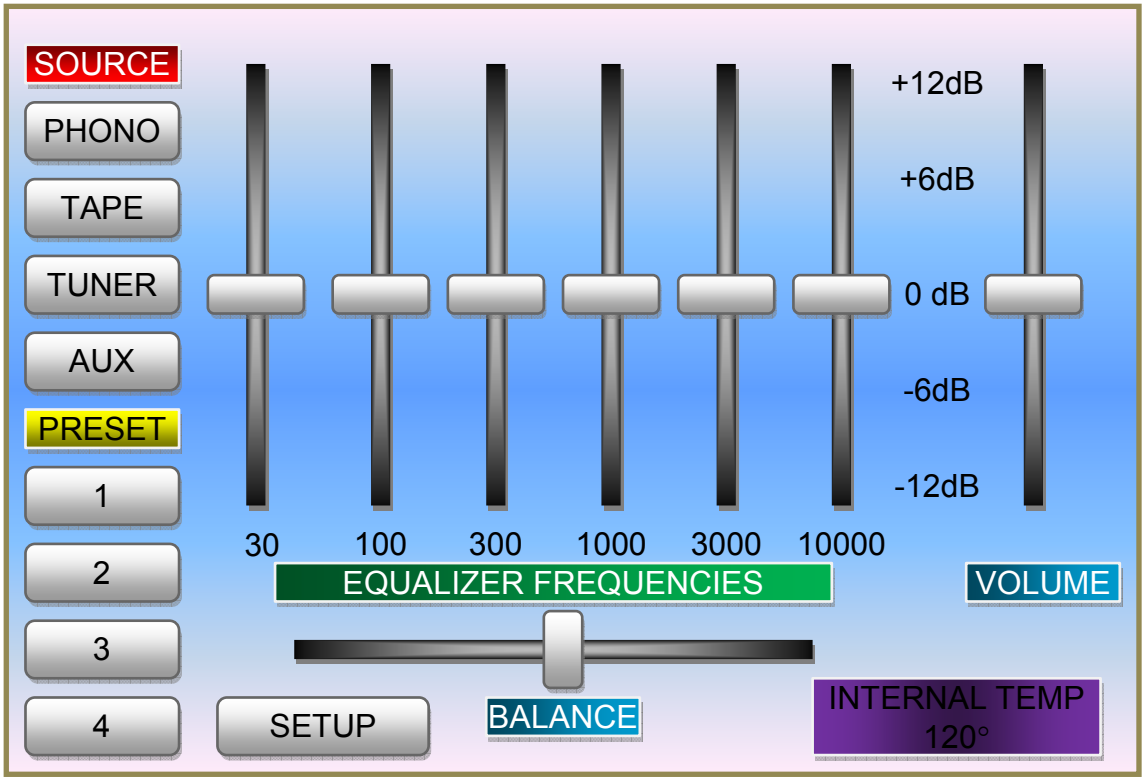


Figure 5: Typical control mode screen.
(Created by the authors)

3: Research Related to Project Definition

After performing an in-depth search of the market it was determined that we had a one-of-a-kind idea. While vacuum tube audio amplifiers are commercially available as will be shown later, it is the LCD / touchscreen interface and graphic equalizer that makes this project unique is that there has never been a digital equalizer interfaced with a vacuum tube amplifying circuit. This may be because vacuum tubes are vastly inefficient compared to their semiconductor counterparts.

3.1: Existing similar projects and products

As a digitally-controlled vacuum tube amplifier this project proves to be unique. We were able to find projects that pertained to portions of our project only after a demanding initial research phase.

3.2: Senior Design Projects from Prior Years/Other Schools

There were a few groups of engineering students that have completed projects containing certain aspects of our projects. We will be using some of their findings to help point us in the right direction when designing and prototyping our project

3.2.1: Low-Cost Audio Power Amplifier

This report was written by a group of undergraduate electrical engineering students in 2005. Even though this project is dated, they still excised good engineering practices and produced valid results. Their abstract states and their goals are as follows:

"The objective of the Fall 2005 Senior Design Project is to design an audio power amplifier that uses no integrated circuits. The team of four engineers was given only one technical constraint; no integrated circuits could be used. However, the team was required to develop other technical constraints that would make the overall design of the amplifier suitable for the "do-it-yourself" market.
The ultimate goal of the team is to design and construct a low-cost audio power amplifier that would appeal to both hobbyists and amateurs alike who are trying to learn about audio amplifiers, but don't want to spend a lot of money, while retaining comparable quality as more costly amplifiers. Our team presents a solution to this problem by providing a 20-Watt Class AB MOSFET audio power amplifier that uses nearly half the number of transistors used in a traditional 20-Watt audio amplifier design. The amplifier design includes a differential pair input stage, a voltage amplifier gain stage, and a MOSFET Class AB output stage. The results produce a power gain of approximately 169dB, and an output power of approximately 20 Watts into an 8-ohm load. The voltage gain is flat for a frequency range of 50Hz to 20 kHz and there is only a 0.08% discrepancy at this range." [28]

Our project has very different technical constraints and many differences. We will be using their decisions on their equalizer circuit and possible look at their power supply circuits for reference.

3.2.2: 10-Band Graphic Equalizer

This project was done by a group of undergraduate engineering students at California Polytechnic State University in 2011. This project mirrors our project in basic functionality. It has a graphical display of the peak outputs of each frequency band, and it also has a fully adjustable 10 band equalizer which is

similar to the six band equalizer that we will be building, but there are many differences between their project and ours. Their abstract is as follows.

"This project consists of a 10-band, two channel graphic equalizer with a light emitting diode (LED) display. Left and right channels are operated on separately by ten sliders for each side. Moving the sliders up or down strengthens or weakens the energy in the corresponding frequency band. The sliders constitute part of an inverting band-pass filter that adds gain or attenuation over a certain range of frequencies, but has no effect outside the band. Each block of the equalizer circuitry acts on one of the ten frequency bands; an inverting summer with unity gain recombines the individual signals. The output of the inverting summer serves as both the speaker and display circuitry input so that the LED display will reflect any signal alterations. A repeated combination of a band-pass filter and peak detector for each frequency band comprises the display circuitry. The band-pass filter used here has a higher quality factor than those used in the equalizer, and removes any content outside the desired frequency band. Once it passes through the filter, the signal continues to the peak detector, where a capacitor follows the signal voltage until a maximum occurs. The capacitor holds the maximum until either a new maximum occurs or the transistor switches on and provides a discharge path to ground. An Atmega328P microcontroller samples the peak detector values of all ten bands and translates the recorded value to a number of LEDs to turn on. The microcontroller runs a real-time operating system (RTOS), using a separate task to sample each frequency band and write related data. After determining which LEDs to light up, the microcontroller transmits the corresponding data to a set of LED display drivers. These drivers handle all display details, including pulse-width modulation (PWM). A 550 watt ATX computer power supply provides all power to the system.
Moving the equalizer sliders modifies whatever audio signal enters the circuit. With the ranges and frequency bands available, the system can negate audio distortion effects such as microphone response, instrument pick-ups, loudspeakers, and room acoustics. The system can also alter the signal to make it sound as if it were recorded or played in a particular environment or way, such as a large hall or over old-time radio"[27]

3.3: Modern Commercial vacuum Tube Amplifiers

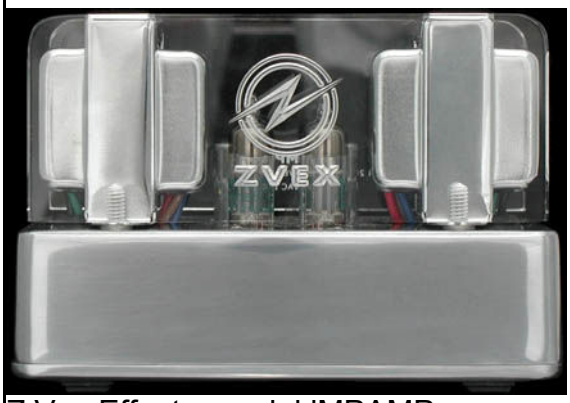
When research for commercially-made products was undertaken, the authors were astounded by the quantity and variety of vacuum-tube based amplifiers available. The selection to be discussed below is a mere sample of what is available. It is perhaps unfortunate that the general public is unaware of them, and even most audiophiles might not know that they exist. Of course, there is still a certain amount of vacuum-tube based equipment likely available in thrift shops or in the dusty corners of countless attics waiting for the day they will either be offered in a yard sale or discarded. Those devices were all made in the glory days of vacuum tubes and will be thirty or more years old. They are not included in the research.

The products to be discussed are available today. They are of modern design and construction, and of high audio quality, comparable to that provided by solid state amplifiers. Due to their prices, they are not available in popular consumer stores such as Best Buy, but must be purchased from dealers that cater to high-end audio products. See Table 3 for a summary of their specifications.

Table 3 Comparison of vacuum tube amplifier models reviewed

JE-Audio, model VM60
(shown without optional tube cover)
Reprinted with permission from John Lam of JE-Audio [4]

Specifications per manufacturer's data:	
Size (H x W x D)	4.7" x 12.8' x 15.6"
Weight	45 pounds
Power output	60 watts
THD	<1%
S/N Ratio	85dB
Bandwidth	20Hz to 20kHz
Price: Not quoted by the manufacturer, but listed as $6300 per pair in The Audio Beat, December 13, 2012 issue.	



Z.Vex Effects, model iMPAMP
Reprinted with permission from Zack Vex of Z.Vex Effects [20]

Specifications per manufacturer's data:	
Size (H x W x D)	3.3" x 3.0" x 3.3"
Weight	Not given
Power output	1 watt per channel
THD	Not given
S/N Ratio	> 80 dB
Bandwidth	10Hz to 20kHz +0/-2 dB
Price	$699

JJ Electronic, model JJ332
Reprinted with permission from Julia Jurcova of JJ Electronic [11]

Specifications per manufacturer's data:	
Size (H x W x D)	10.3" x 22.4" x 15.0:
Weight	92 pounds
Power output	20 Watts per channel
THD	Not given
S/N Ratio	Not given
Bandwidth	18 Hz to 25 KHz +0/-1 dB

Price: Not quoted by the manufacturer, but listed as $6000 on the Divergent Technologies (dealer of high-end audio products) web site

3.4: Component Research

Initially it was planned to enter our design in the 2012-213 Texas Instruments US and Canada Analog Design Contest [6], which requires the use of three parts made by Texas Instruments, two of which must be analog. Since our design is vacuum-tube based, the only analog parts that we could buy from Texas Instruments (TI) were analog multiplexers, operational amplifiers, voltage regulators and the microcontroller. Doing so enables the savings of up to $200 in costs for the project because TI provides a coupon code for ordering. Late in the first semester it was discovered that the MCU on the Stellaris Launch Pad is a new part that is not yet available for sale. As a result, since TI did not have a suitable alternate MCU, one from STMicroelectronics was chosen as a replacement. Since the TI MCU will no longer be used, it was decided that our project will be withdrawn from the competition.

3.4.1: Vacuum Tubes

Virtually all electronic equipment manufactured since the late 1970's uses transistors and/or integrated circuits. It would seem that the era of vacuum tubes has passed, "(h)*owever, tubes still find uses where solid-state devices have not been developed, are impractical, or where a tube has superior performance, as with some devices in professional audio and high-power radio transmitters*." [7]. One of those applications is for electronics intended to be used in the harsh environment of outer space, and indeed, vacuum tubes are poised for somewhat of a comeback: "*The new device is a cross between today's transistors and the vacuum tubes of yesteryear. It's small and easily manufactured, but also fast and radiation-proof. … the 'nano vacuum tube,' … is created by etching a tiny cavity in phosphorous-doped silicon. The cavity is bordered by three electrodes: a source, a gate, and a drain. The source and drain are separated by just 150 nanometers*." [3]. This is a very interesting discovery! However, this project requires vacuum tubes that are currently available.

No American company still makes vacuum tubes. Remaining production has long ago shifted overseas to companies in Russia and generally Eastern Europe countries: Sovtek, Tung-Sol and JJ Electronics are a few examples (all three companies are subsidiaries by New Sensor Corporation, itself located in New York City). One vendor of vacuum tubes located is Thetubestore.com. Given their vast selection, ready availability and reasonable prices of vacuum tubes, they were selected early as the vendor of choice [18].

Hundreds of part numbers of vacuum tubes were designed during their heyday in early to mid-20th century, and most of the part numbers are indeed still available, either as New Old Stock (made decades ago, but never used) or of brand new manufacture. During research of the types of vacuum tubes to use in this project, certain part numbers tended to appear frequently, with one in particular, the type

6L6. This tube is commonly used in amplifier designs intended for the hobbyist market, so a number of reference schematics were available. The trade study also required that a PSpice model of the tube, which have been created for only a limited number of tubes of which 6L6 is one. The catalog page on Thetubestore.com's website lists 11 entries for the 6L6, made by seven different companies, most of which are in stock. Hence, the type 6L6 was chosen as the power amplifier tube.

The architecture of the power amplifier requires a phase splitter. Any of a number of tubes could have been chosen but using a dual-triode has the advantage of requiring one less tube to buy. During the trade study to be discussed in the following sections the type 12AU7 tube was used initially, but during the design refinement, a 12BH7 type was found to offer slightly lower distortion characteristics. Since the price of all tubes in the 12(letters)7 type of tube are comparable, and PSpice models of both tubes are available, the type 12BH7 was chosen as the phase splitter tube.

Historical note: The 6L6 vacuum tube was invented in 1936 and has remained in continuous production since then. That makes the 6L6 the longest continuously manufactured electronic device, if not of any product, of all time. [7]

3.4.2: Display Unit

The Initial Project Document specifies an LCD Display. LCD Displays have the quality necessary to show the user interface and display animated graphics that move along with the music, and have substantially decreased in price in recent years. The process of choosing the right LCD Screen, however, took several weeks. Some of the aspects taken in consideration were:

Table 4 LCD Requirements

Item	Requirement
Panel dimension	not to exceed 10" x 10" x 3"
Screen Dimension	6" to 7" diagonal
Refresh rate	at least 50 milliseconds
Image quality	minimum 8 million colors
Cost	not to exceed $100
Interface	digital
Documentation	ensure sufficient documentation is available
Availability	Ability to receive product within 30 days of purchase

It was decided that six to seven inch diagonal displays were sufficiently large to be able to display equalizer controls and animated graphics, be large enough to view the visualizations from across a room, yet be small enough to be mounted

on the front surface of the final product. A touch screen was decided from the beginning as the user method of input and as the truly innovating feature of the project since no commercial tube amplifier offers touch screen interface.

After a detailed comparison and a discussion between all team members, a unanimous decision was made to use the *7" inch TFT 800*480 LCD Display Module 16M colors Touch Panel Screen SSD1963* for the project. This panel meets all of the design requirements, and provides additional features.

The LCD is controlled by an integrated Solomon Systech SSD1963 Controller and be controlled by the Microcontroller Unit CCA. However, most of the refreshing and LCD controlling will be performed by the integrated controller.

The Touch Screen input is obtained via an integrated Shenzhen XPTEK Technology XPT2046 Controller. This input is handled by the MCU CCA. In addition, this Display Panel offers a micro-SD card slot which could be used to store background images and control panel graphics.
The factors that lead the team to choose this device to be used as the User Interface with the amplifier to control the amplifier as well as graphic display of musical visualization are shown in Table 5

Table 5 [22]

Factor	**Detail**
Cost	$57.29
Screen dimension	7” diagonal
Panel dimension (width x height x depth)	6.38” x 3.79” x 0.67”
weight	0.0445Kg
Resolution	800x480
Colors	16 million
LCD Controller	SSD1963 (integrated)
LCD Controller documentation	available
Touch Screen controller	XPT2046
Touch Screen controller documentation	available
Availability	within 20 days

The following description and list of features (see Table 6) were gathered from the documentation file of the integrated LCD Controller SSD1963 Product Preview file provided by vendor of panel.

“*SSD1963 is a display controller of 1215K byte frame buffer to support up to 864 x 480 x 24bit graphics content. It also equips parallel MCU interfaces in different bus width to receive graphics data and command from MCU. Its display interface supports common RAM-less LCD driver of color depth up to 24 bit-per-pixel.”* [23].

Table 6 Features [23]

Display Features	Built-in 1215K bytes frame buffer. Support up to 864 x 480 at 24bpp display
	Support TFT 18/24-bit generic RGB and TTL interface panel
	Support 8-bit RGB interface
	Programmable brightness, contrast and saturation control
MCU connectivity	8/9/16/18/24-bit MCU interface
	Tearing effect signal
I/O connectivity	4 GPIO pins
	Built-in clock generator
	Deep sleep mode for power saving
	Core supply power (VDDPLL and VDDD): 1.2V±0.1V
	I/O supply power(VDDIO): 1.65V to 3.6V
	LCD interface supply power (VDDLCD): 1.65V to 3.6V”

Since the controller is already integrated into the panel, we do not need to worry about how the controller is connected to LCD display. However, we will need use the controller’s documentation to design an interface write the software for the final product.

The integrated Touch Screen controller is the XPT2046 manufactured by Shenzhen Xptek Technology CO, LTD. The following description and features have been gathered from the documentation provided by vendor in the XPT2046 Touch Screen document. [24]

“The XPT2046 is a 4-wire resistive touch screen controller that incorporates a 12-bit 125 kHz sampling SAR type A/D converter.[In addition, it] operates down to 2.2V supply voltage and supports digital I/O interface voltage from 1.5V to VCC in order to connect low voltage uP. [It] can detect the pressed screen location by performing two A/D conversions. In addition to location, the XPT2046 also measures touch screen pressure. On-chip VREF can be utilized for analog auxiliary input, temperature measurement and battery monitoring with the ability to measure voltage from 0V to 5V. The XPT2046 also has an on-chip temperature sensor.” [24]

Some of the important features of the XPT2046 controller that is integrated in this panel which are of matter to this project’s planning and development are shown in the following table.

Table 7 [24]

4-wire interface	
Sampling frequency	125 KHz (max)
Pen pressure measurement	
Thermo sensor	on-chip
Operating temperature	-40°C to +85°C

3.4.3: Embedded Processor

The embedded processor on the Microcontroller CCA handle all the digital signal processing, calculations, communications, graphics updates in the system, and startup timing. In order to be able to execute all of these needed functions in a reasonable amount of time, a microcontroller which has a clock rate of greater than 50 MHz was required. To integrate all the digital components such as the digital resistors, graphics display unit, and the touch screen we will need a minimum of 35 general purpose input output pins. In order to create a real time visual of the analog audio input signal we will need digital signal processing functionality, such as analog to digital convertors, and floating point capabilities. Due to the fact that a large number of functions will be happening in real time the project requires an adequate architecture for deal with interrupts. The selected controller must have some sort of development board available to ease of prototyping and software development. Finally, since this project was self-funded cost was important.

With these characteristics in mind we began our search for a microcontroller for our project. In our research we considered a number of different controllers, which must meet the requirements of Table 8.

Table 8:

Microcontroller Requirements	
Hardware DSP capability	Required
Hardware MAC	Required
Floating Point UInit	Required
Minimum clock rate	50 KHz
Data flash size (KB)	128 kilobytes
Program flash size	128 kilobytes
SRAM size	512 kilobytes
Serial port interface	I^2C or SPI ports
On chip debugging	Required
GPIO (mim)	35
DSP library	Required
Premium dev environment	Required
Analog input pins	2
minimum ADC resolution	10 bits

DMA	Required
interrupt control system	Required
Active development	Required
Development board available	Required

3.4.3.1: Atmega2560

The Atmega2560 was considered for the project due to the vast amount of community support. The Atmega2560 is the microcontroller that is incorporated in the Arduino Atmega2560 development board. The Atmega2560 meets most of our specification requirements. At 256 Kbytes of flash it has plenty of memory for our code. The max operating frequency is 16MHz which is a little slow to meet our estimated clock frequency. The CPU is an 8-bit AVR. The Atmega2560 exceeds our required GPIO, with 86 pins. It implements 32 external interrupts, more than sufficient. It has 16 analog-to-digital conversion (ADC) channels at 10 bit resolution per channel. A large negative is that it does not have a FPU which we require to do real time Fourier transforms. The most popular IDE for the Atmega2560 is the Arduino IDE. There is a vast number of preexisting libraries for all sorts of applications in this IDE. The code can be written in a form of the C-programming language. A downside to the using the Arduino IDE is that it is made for beginners and adds a level of difficulty when trying to perform low level operations. Most of the libraries are written with the average hobbyist in mind and most of the functionality is centered on robotics and sensor interfacing. [29]

3.4.3.2: Stellaris LM4F120H5QR

The Stellaris LM4F120H5QR was the second microcontroller that we investigated for our project. The Stellaris is an ARM-based processor produced by Texas Instruments. TI provides a vast amount of support for this chip and there is an entire suite of development tools, and APIs written to ease development on the Stellaris called StellarisWare. StellarisWare has included a full basic graphics library; very attractive since we will be integrating a 7” LCD display. The Stellaris has the following features that interest us: With 256KB of Flash it has the same as the Atmega2568 and meet our requirements. The max operating frequency of 80MHz exceeds our requirement. The CPU is the ARM Cortex M4F which is a 32-bit RISC based processor. The Stellaris meets our GPIO requirement in the H5QR package which has 43 pins. It has an impressive specially-design system to deal with interrupts called “integrated Nested Vectored Interrupt Controller (NVIC)” offering an easy and efficient way to deal with large numbers of simultaneous interrupts. Another nice feature that was included in the Stellaris was the “IEEE754-compliant single-precision Floating-Point Unit (FPU)” which will enable us to do real time Fourier transforms. In the analog field the Stellaris performs very well. With 12 ADC channels at 12 bit resolution capable of sampling at 1000kSPS makes the Stellaris well equipped to

handle all of the analog conversions and manipulations required. When it comes to programming and developing for the Stellaris TI makes it easy by providing many options to pick from including TI's proprietary IDE, Code Composer Studios (CCS). CCS is a plug-in to the well-known IDE Eclipse which is a professional grade IDE. Through CCS you are given full access to all the low under lying subroutines that we will need to interface with. TI sells a special launch pad development board for the Stellaris LM4F120H5QR which will aid us when it comes to prototyping. After review the Stellaris appeared to be a possible good fit to act as the microcontroller. [30]

3.4.3.3: MSP430

The Texas Instruments MSP430G2553 was the third microcontroller that we investigated for our project. TI has a large amount of preexisting libraries that provide a vast amount of support for implementing various functions on the MSP430. There is a complete API available called MSP430WARE provided by TI. MSP430WARE provides functionality such as I^2C and other serial interfaces, analog to digital conversion, and much more. The MSP430 is a based on a 16-bit RISC architecture. The max operation frequency of the internal main clock on the MSP430 is 16 MHz, which is too slow for our project. The 24 GPIO available does not meet our requirements. It has just 16KB of flash memory which may be enough for our project but it would be cutting it close. With regards to analog to digital converters it has 8 channels at 10 bit resolution and meets our requirements. Another draw-back is that the MSP430 has no floating point unit making it impossible to do the Fourier transforms that we are requiring. After review the MSP430 was deemed not be a good fit to act as the microcontroller and was eliminated from consideration. [31]

3.4.3.4: STM32F303VCT6

The STM32F303VCT6 is made by STMicroelectronics and was the final microcontroller reviewed. It is part of the STM32F3xx series which is an ARM M4 core based microcontroller family. The STM32F3xx series packs a lot of great functionality and power on a single chip. The ARM M4 core is top of the line core and its max CPU clock rate of 72MHz puts it into the high speed class. It is easily fast enough to carry out all the routines that are going to be required of it. The STM32F3xx series employs a single cycle multiplication and division hardware design which will greatly reduce the number of clock cycles needed to perform the FFT that is going be running all the time. Another great feature is a hardware-implemented DSP supported by a FPU and MPU that offers great flexibility. The unit comes with 256KB of flash memory and 40KB of SRAM, both of which should be plenty for full implementation of the project design. The power requirement is a standard 3.3 volts maxing out at a 160 mA draw. There are a few different package designs that the STM32F3xx series comes in, each with different amounts of GPIO varying from 48 up 100 pins. After full system design an exact amount of GPIO will be known and will allow the selection of an

appropriate package. The STM32F3xx series has all the required features with DSP, ADC, DMA and JTAG. There is a development board available with the same part number chip on it and allows access to all the major functionality offered by the chip. The development board also offers out-of-the-box functionality allowing for fast prototyping and speedy circuit design. Also STMicro supplies a very user friendly integrated development environment. Based on the above analyses the STM32F303VCT6 turned out to be a great candidate for the microcontroller. [32]

3.4.3.5: Embedded Processor Final Decision

After a brief market analysis there were two candidates to be considered: TI's ARM-based Stellaris series, and STMicroelectronic's ARM-based ST32F3 series. Both of these controllers come in a wide variety of packages each of which offers different characteristics. STM and TI both supply development boards for both of these controllers. In fact, both are nearly identical. During the first part of the design process the team had entered the TI design contest there was a favor for the Stellaris, and development boards where purchased quickly. Only after further review and after designs had already been created with the Stellaris issues occurred. The Stellaris' development boards where obtained on the assumption that the silicon was available, but this is where problems occurred: It turned that the TI Stellaris parts were unavailable for purchase despite the fact that TI distributes the development boards. As a result the final decision was made to go with the STMicroelectronics STM32F303VCT6.

The chosen microcontroller contains a 32-bit ARM Cortex M4 core that provides high quality digital signal processing, floating point calculations, and many advance peripheral functions at a clock rate of 72 MHz. The STM32F3 series is a full system-on-a-chip solution for saving space on printed circuit boards and is optimized for efficient handling and processing of mixed signals in circuits such audio filters. Table 9 lists some of the features of STM32F3 series processors have that apply to this project. [32]

Table 9

Features		Benefits
Performance	72 MHz/62 DMIPS (from flash) or 94 from CCM-SRAM*)	Boosted execution of control algorithms
		More features possible for your applications
		Ease of use
	Cortex-M4 with single cycle DSP MAC and floating point unit	Better code efficiency
		Elimination of scaling and saturation
Features		**Benefits**
Real-time performance	DMA controllers	More performance for critical routines with zero-wait state execution from safe CCM-SRAM
	Memory protection	
Maximum integration	Up to 256 Kbytes of on-chip Flash memory, up to 48 Kbytes of SRAM, reset circuit, internal RCs, PLLs,	More features in space-constrained applications
Superior and innovative peripherals	Analog: 4x 12-bit ADC 5 MSPS* reaching 18 MSPS in interleaved mode, 3x 16-bit sigma-delta* ADC up to 50 KSPS	Full set of integration features on chip resulting in simplified board designs and fewer external components
	Up to 17 timers: 16 and 32 bits running up to 144 MHz	BOM cost reduced
	Up to 12 communication interfaces	Digital signal processing (DSP) capability at competitive price
	Cyclic redundancy check	

3.5: Operational amplifier

Due to the TI design competition, all other operational amplifier vendors were ruled out. Using TI's online part selection process (Products for Audio Operational Amplifier), it was quickly learned that TI makes high-performance operational amplifiers intended for audio processing applications, ruling out standard op-amps such as the TL081. Modern op-amps can be considered essentially perfect, so only the following parameters were considered: Noise voltage (V_n), THD, Price, availability in a DIP, and model support in NI Multisim. It was quickly determined that the LM4562NA would be an overall excellent choice. This part was used in the graphics equalizer circuit.[14]

3.6: Analog Multiplexer

Due to the TI design competition, it would have been preferable to have used as many TI parts as possible, but TI's best offering was not a good fit for our requirements as will be shown. The requirement is for four AC input sources to be selectable with the highest isolation between channels available in a DIP. Since the physical implementation will be two identical CCAs, one per channel, a dual-channel switch was not considered. Four parts were considered, and are summarized in Table 10. All parts are available in a DIP, have control voltage ranges compatible with the MCU, and are available from Digi-Key. Despite TI part's much-lower price, the Analog Device's AD8184ANZ part excels in every rating and would seem to be the ideal choice. Upon closer inspection it was realized that the AD8184ANZ is designed for video up to 700MHz and has specific circuit layout concerns. Some simulation was performed using it in a test circuit and it proved very troublesome. Because of these issues, the ADG408BN was be used in the input source select circuit.

Table 10: Specification comparison of parts for the input source select circuit. All values are from the respective manufacturer's data sheet.

Parameter	Texas Instruments SN74LV4051	Analog Devices AD8184ANZ	Analog Devices ADG408BN	Maxim DG508
Configuration	8:1	4:1	8:1	8:1
Supply range	–0.5 V to 7 V	±4V to ±6V	±5V to ±15V	±4.5V to ±18V
Input range	0 to 7 V	$\pm V_{SUPPLY}$	$\pm V_{SUPPLY}$	±15V
Crosstalk	-45dB	-98dB	-85dB	-68dB
Noise Voltage	Not rated	4.5 nV / √Hz	Not rated	Not rated
THD	Not rated	-74dBc	Not rated	Not rated
$R_{DS(ON)}$, max	225Ω	Not rated	125Ω	450Ω
Cost, each	$0.17	$5.75	$6.15	$6.31
Special switch feature	No	No	Break before Make	No latch up

3.7: Digital Potentiometer

The project will be using digital potentiometers in the 6 band equalizer, volume controller. The digital potentiometer allows us to incorporate digital control into an analog signal path. The requirements for the digital potentiometers are as follows.

- I^2C or SPI compatible
- 3-4 bit address
- 128 or 256 possible resistor values.
- Available on a PDIP
- Signal range of a minimum of ± 2.5V

- Low THD
- Dual supply
- Low cross talk if 2 channel

Serial compatibility is required due to the limited number of pins available to interface with the microcontroller and for a simpler design and layout. Two serial busses were considered, SPI or I^2C, and will be discussed later in this document.

The CCAs shall be designed such that a digital potentiometer on channel A can share the same address or chip enable line as the corresponding digital potentiometer on channel B. With this observation the amount of different address can be reduced from 16 to 8. The reduction is needed because addresses enables the use of digital potentiometer with a minimum of 3 address bits if the decision is made to use I^2C.

The input signal is zero-centered and up to 2.5 Vpp. Because of the negative component of the signal a digital potentiometer with dual sources is needed. A part with the lowest possible noise, THD and crosstalk is desired, because any such effects will be amplified by later stages. There is no requirement for the potentiometer to store the “position” of the potentiometer because they will be reset to a certain value upon power-up. Analog Devices offers an excellent selection of digital potentiometers, and using their online part selector, several candidate parts were quickly located. A 10K ohm value was chosen from the very limited selection of values because only the voltage division ratio is important in the circuits and a higher value is preferable to minimize signal loading. For ease of ordering, all potentiometers will have the same value. The choice of parts was narrowed to the Analog Devices AD5160 and AD8400 series, with the AD8403 ultimately chosen because of DIP availability.

To decrease the time needed for prototyping and to decrease the complexity Dual In-line Package (DIP) parts are desired wherever possible. Using DIP parts for the digital potentiometers (as well as for other parts) reduces the price because the same components that were used for prototyping were reused on the final circuit boards. This decision turned out to be a wise choice because the selected Analog Devices AD8403 chips, while offering high performance, proved to be extremely sensitive to electrostatic discharge (ESD) and incorrect voltages. As a result, this part proved to be very troublesome and because of many mishaps resulted in damaged parts. Eight of this part number were thusly damaged and had to be replaced, the most of any part during the entire project. Having insisted on DIP parts made their replacement and troubleshooting much easier.

3.8: Serial Communication Bus

The project will be using a serial bus to support all the digital components in the project. With the current design the only digital components being used are digital potentiometers, sixteen in number. There were two possible serial protocols considered for the project which are Internal-Integrated Circuit **(I^2C)** and Serial Peripheral Interface (**SPI)**. Both protocols are capable of performing the needed communications with the digital potentiometer. To make the final choice, the following aspects were considered.

- Difficulty to implement
- Number of pins needed for implementation
- Reliability
- Speed
- Number of available addresses

When it comes to difficulty of implementation both protocols pose their own challenges. With I^2C the difficulty lies in the software programming aspect of the protocol. With I^2C, slaves on the bus have the ability to hold the clock pulse high if the master is reading from it. This is because the master may be driving the clock faster than the slave can put the data on the line. This problem may turn out to be an issue if the implementation to check the value in the potentiometer's registers is later desired. Also with I^2C there can only be $2^{\text{Slave address bits}}$ units on the same address line. This can pose a problem because it is hard to find a digital potentiometer that meets all the needed analog requirements and also has an acceptable amount of address bits. For the project a unit would have to at the minimum 3 address lines in order to meet the minimum required amount. Then there is SPI which does not have a strict protocol to implement such as specific word size per transmission. SPI is also fairly easy to implement in hardware.

Given the choice of STM32 series processor, the number of GPIO pins required to implement either bus is small and not an issue. Since the digital potentiometers are being used in a sensitive circuit, the potential for noise is a concern. Due to the fact that I^2C is edge-driven this makes it more prone to noise interference which could possibly cause missed data bytes. Also because each node must be addressed via 3 to 4 bits this means there is an extra 4 bits that must go out just to begin communication and then the register address that data is being accessed from has to be sent. All of this addressing decreases the overall throughput of the bus. In order to maintain the same throughput the clocking frequency must be increased which increases the probability of errors and also increases the noise to the analog circuitry. With SPI there is no on bus addressing required because each chip is either on are off via the chip select line. The only transactions that need to happen are a 0 or 1 sent to each node to select / deselect it followed by the desired data bits which are sent out on the Master Out Slave IN (MOSI) line. Each bit gets clocked in from the MOSI line on every specified edge of the clock. Because SPI is level triggered and not edge triggered its less prone to errors do to noise from the analog circuitry. Also because there is less information being transmitted a higher throughput can be

achieved at much lower frequencies which decreases the noise that could be coupled to the analog circuitry. With SPI the data signals are unidirectional so optocouplers can be used to galvanicly isolate the digital and analog circuits.

After performing an in-depth market search it turns out that there is a far wider selection of SPI enabled digital potentiometers meeting the requirements versus those that are I^2C enabled. Table 11 below summarizes all the differences between I^2C and SPI.

Table 11

Protocol	**SPI**	**I^2C**
Pins needed to fully implement	6	2
hardware implementation available	Yes	Yes
Data transfer type (edge / level)	Level	Edge
Required time complete one full refresh all digital potentiometer values @ 10KHz (transmit 96 bits)	.8 mS	1.3 mS
Capable of being galvanic isolated	Yes	No
Pull up resistors required	No	Yes
Difficult to find Parts	No	Yes

Based all the findings in Table 11 the decision was initially made to use the SPI protocol over I^2C, as it offered higher throughput, more reliability, and better protection from noise. Ultimately however, the SPI bus was abandoned in favor of a simpler bit-banger approach due to an SPI bus being overkill, as no other devices are on the bus other than the digital potentiometers.

3.9: Optocoupler

Optocouplers are required to solve three problems: 1) Create galvanic isolation between the digital and analog processing and ease grounding difficulties, 2) Prevent noise generated by the MCU from being coupled into the analog circuits, and 3) Bias the SPI bus signals to be compatible with the bipolar requirements of the digital potentiometers. There are several factors that greatly limit the choice of parts: First and foremost is the drive capability of the GPIO and second is availability in a DIP. According to the STM32F303-series spec sheet, the GPIO provides an upper limit of 20mA maximum. As a result ordinary optocouplers cannot be used due to their >50mA drive requirement. (To minimize costs and circuit complexity, direct drive from the microcontroller would have been preferable.) However Vishay Semiconductors manufactures a dual optocoupler series with a very low 1.6 to 5mA drive requirement. The dual nature means

fewer parts are required, thus the Vishay SFH6731 was chosen as the optocoupler.

3.10: Cabinet Materials

The key parameters are aesthetics, workability and cost. Ideally a dense, dark type of wood such as walnut would be preferable. Each panel must consist of a single board to minimize construction effort, but specialty woods are very expensive in the size shown in Figure 2. Thus, the front and side panels will be fabricated from 12" pine plank purchased from Home Depot and stained to simulate walnut. For the metal sheets, some means to allow ventilation is required and the easiest way to do that is to use a heavy gauge of perforated sheet metal. Perforated sheets are available in a variety of metals from vendors catering to hobbyists but are somewhat expensive. Thus the top, bottom and back surfaces will be fabricated from scrap metal. Skycraft will be investigated as the source. Metal surfaces will be iridized black (or simply primed and painted if iridizing proves costly). A type of high permeability metal such as steel, Permalloy or mu-metal is required for internal shielding. The ideal choice would be relatively soft to allow easy working. Ultraperm 80 Metal Shield from 3M will be investigated for this purpose.

3.11: Critical Components

Critical components are defined as those that would have put the successful completion of this project in jeopardy if a suitable candidate could not be located, the parts failed to arrive from the vendor in time to support fabrication and test, or they would require excessive lead time to replace should they become damaged. The list of critical parts is given in Table 12.

Table 12

Component(s)	Status
Vacuum tubes	12BH7A and 6L6 These are very fragile and prone to damage
Impedance matching transformers	Hammond Manufacturing 125E
High-voltage transformer for vacuum tube plate circuits	Hammond Manufacturing 263CZ. At $67, this is the most expensive single component.
High-current transformer for vacuum tube filaments	This transformer is a very large and heavy 12V / 8A that one of the authors already had. It would be expensive to replace.
LCD / Touchscreen module	7" inch TFT 800*480 LCD Display Module 16M colors Touch Panel Screen SSD1963. This part has a 30-day lead time, minimum.
Microcontroller	ST Microelectronics STM32F303VCT6
Op-amps	LM4562AN op amps

High voltage capacitors	Relatively small units were located from Digi-Key but they turned out to be too small to completely filter the hum. Larger 1000μF / 500VDC capacitors were found in the lab. These would be expensive to replace.
Metal sheets for the cabinet	Source is TBD

3.12: Possible Architecture

In this project there will be several topologies considered for each circuit section. There will be trade studies performed to evaluate each topology.

3.12.1: Vacuum Tube Preamplifier / Amplifier

To aid the choice of design methodology for the vacuum tube circuits, a trade study was performed. This study consisted of designing several circuit configurations on National Instruments Multisim, performing simulations using a 1 KHz signal source, and comparing their performance. The following circuit configurations were considered:

1. Push-Pull vacuum tube without global feedback
2. Push-Pull vacuum tube with global feedback
3. Single Ended Pull vacuum tube without global feedback
4. Single Ended Pull vacuum tube with global feedback
5. Single Ended transistor, without global feedback

“Global feedback” refers to negative feedback made from the output of the speaker terminals all the way back to the preamplifier input. Empirical attempts were made to optimize each design for the lowest possible overall THD. Each design was measured for THD at various output power levels, and analyzed for distortion characteristics, power requirements and frequency response.

Design 1 turned out to be the best overall: A type 12AU7 dual triode tube configured as a phase splitter driving two type 6L6 beam power pentode tubes configured as a push-pull amplifier. This circuit easily achieved 18 watts RMS, resulting in about 2.9% THD. This value is relatively high however the THD appears to be a function of output power, dropping to a relatively modest level of .3% at -18dB. Fourier analysis reveals that the distortion has a dominant second order component, characteristic of the “tube sound” sought after by audiophiles. This circuit had the very best frequency response of all designs, and is essentially flat across the 20 Hz to 20 KHz band.

Design 2 was the same as Design 1 except with a feedback path from the output of the transformer to the input of the phase splitter. This feedback reduced the maximum power. The THD at maximum output was increased, though it was lower than Design 1 at all other powers tested. While these effects are not

worrisome, the feedback caused a pronounced loss of low-end response. Given similar characteristics to Design 1 otherwise, Design 2 was eliminated since it provided no compelling benefits.

Designs 3 and 4 consisted of a type 6J5 single triode tube driving one type 6L6 beam power pentode tube; both tubes were configured as common-cathode amplifiers. Immediately it was discovered that a single-ended design would not meet the output power requirement without undue distortion and thus both were eliminated from consideration. Designs 3 and 4 were slightly more power efficient than designs 1 and 2. The frequency responses were comparably to their push-pull counterparts.

Design 5 was not in the tradespace however it was included in the study to offer a comparison to vacuum tube designs. It should be noted that all designs considered so far are very simple – vacuum tubes simply require less support circuitry than transistors for a given performance level. Thus it was very hard to design and build a simple transistor amplifier that could drive a speaker at comparable power levels to the vacuum tube designs. The resulting design uses a type MJD243G NPN transistor configured as a common-emitter amplifier driving an impedance matching transformer – a topology similar to design 3 (it even requires a relatively high supply voltage). Design 5 offers very low distortion characteristics, though it suffers from very poor low frequency response, and very poor efficiency. The real surprising result was the distortion characteristics: The dominant 2^{nd} and 4^{th} order components might give this amplifier a tube-like sound, suggesting that “tube sound” could be a function of topology.

Based on these tests, the design approach chosen will be the Design 1 push-pull design without global feedback. This circuit forms the “reference circuit” for the hardware design details in Section 4. See Table 13 and Table 14 for a summary of the test results.

Table 13: Trade study simulation results

Design	Topology	Max Vout (Vpk)	Max Power (Wrms)	THD at 1KHz	THD at -6dB	THD at -12dB	THD at -18dB	THD at -24dB	Freq Response	Total DC Power (W)
1	12AU7 phase spitter + 2X 6L6 amp	18	20.24	2.95	1.3	0.61	0.293	0.165	18.3 dB, 20Hz-100KHz down <1dB at ends	141
2	12AU7 phase spitter + 2X 6L6 amp	15.3	14.63	3.8	0.96	0.42	0.27	0.16	18.3dB 500Hz-100KHz down 3dB @ 92Hz	142
3	6J5 preamp + 6L6 amp	9.2	5.29	3.7	1.98	0.4	0.17	0.09	-1.2dB, 20Hz-100KHz down <1dB at ends	28
4	6J5 preamp + 6L6 amp	9.1	5.17	1.9	1.2	0.04	0.017	0.08	-0.5dB, 90Hz-100KHz down 3dB @ 20Hz	28
5	One MJD243G transistor, no preamp	9.7	5.88	0.6	0.4	0.4 (jittery)	0.4 (jittery)	not measured	-9dB, 400Hz-100KHz down **18dB** at 20Hz	400
1	Push Pull Vaccum Tube, no global feedback									
2	Push Pull Vaccum Tube, with global feedback									
3	Single-Ended Vacuum Tube, no global feedback									
4	Single-Ended Vacuum Tube, with global feedback									
5	Single-Ended Transistor, no global feedback									
	All measurements conducted at steady-state									

Table 14: Detailed Fourier analysis results for design 1

Harmonic	Frequency	Magnitude	Phase	Norm. Mag.
1	1000	18.5659	-179.37	1
2	2000	0.428866	90.9477	0.0230996
3	3000	0.170223	-178.41	0.00916855
4	4000	0.0129103	86.7582	0.000695376
5	5000	0.0107462	2.27143	0.000578814

3.12.2: Top Level Interconnect Diagram

Figure 6 provides the top-level interconnect diagram showing all CCAs, cables and input/output connections.

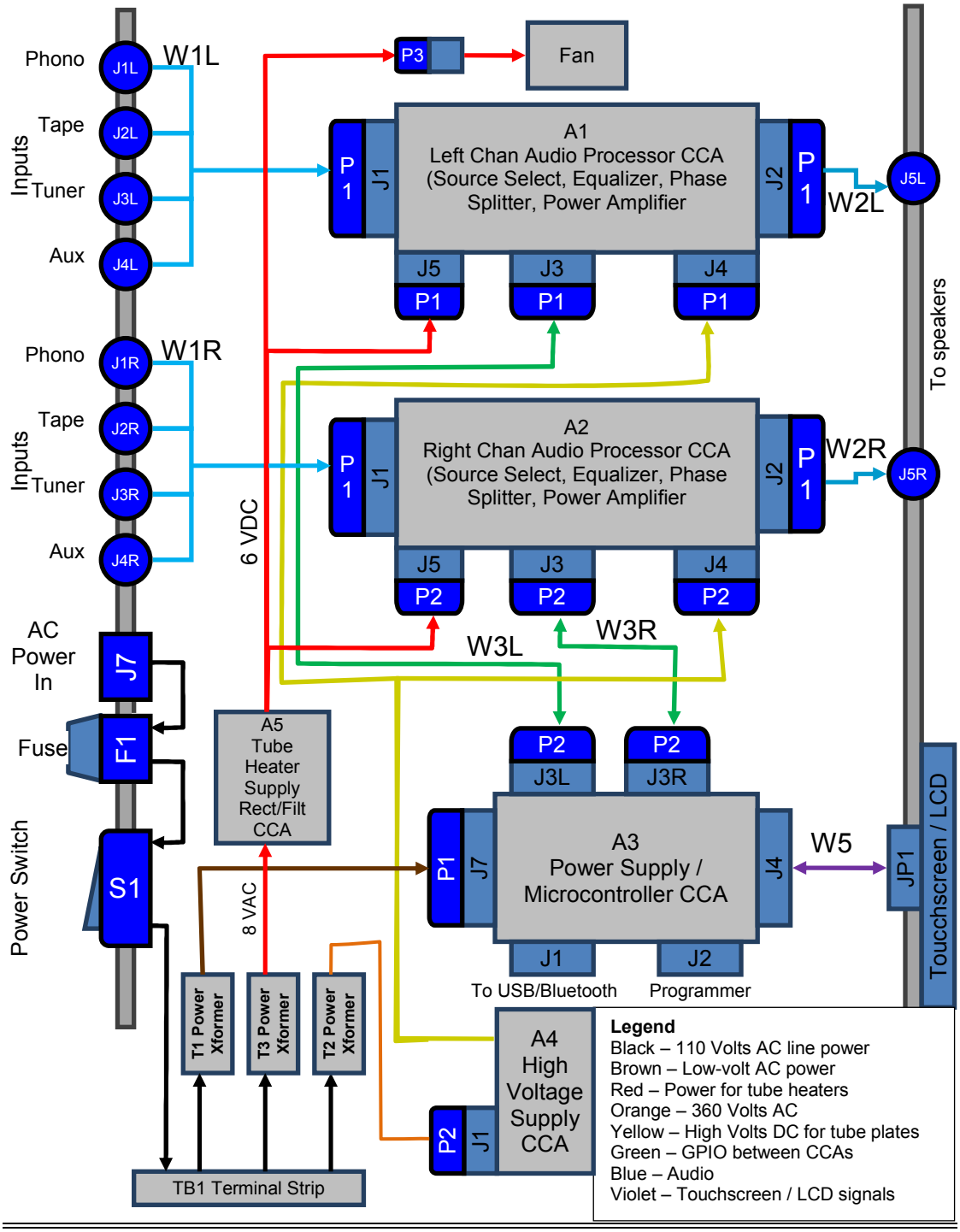


Figure 6: Top level interconnect diagram. (Created by the authors)

3.12.3: Audio Processor Diagram

Figure 7 provides the block diagram of the Audio Processor CCA

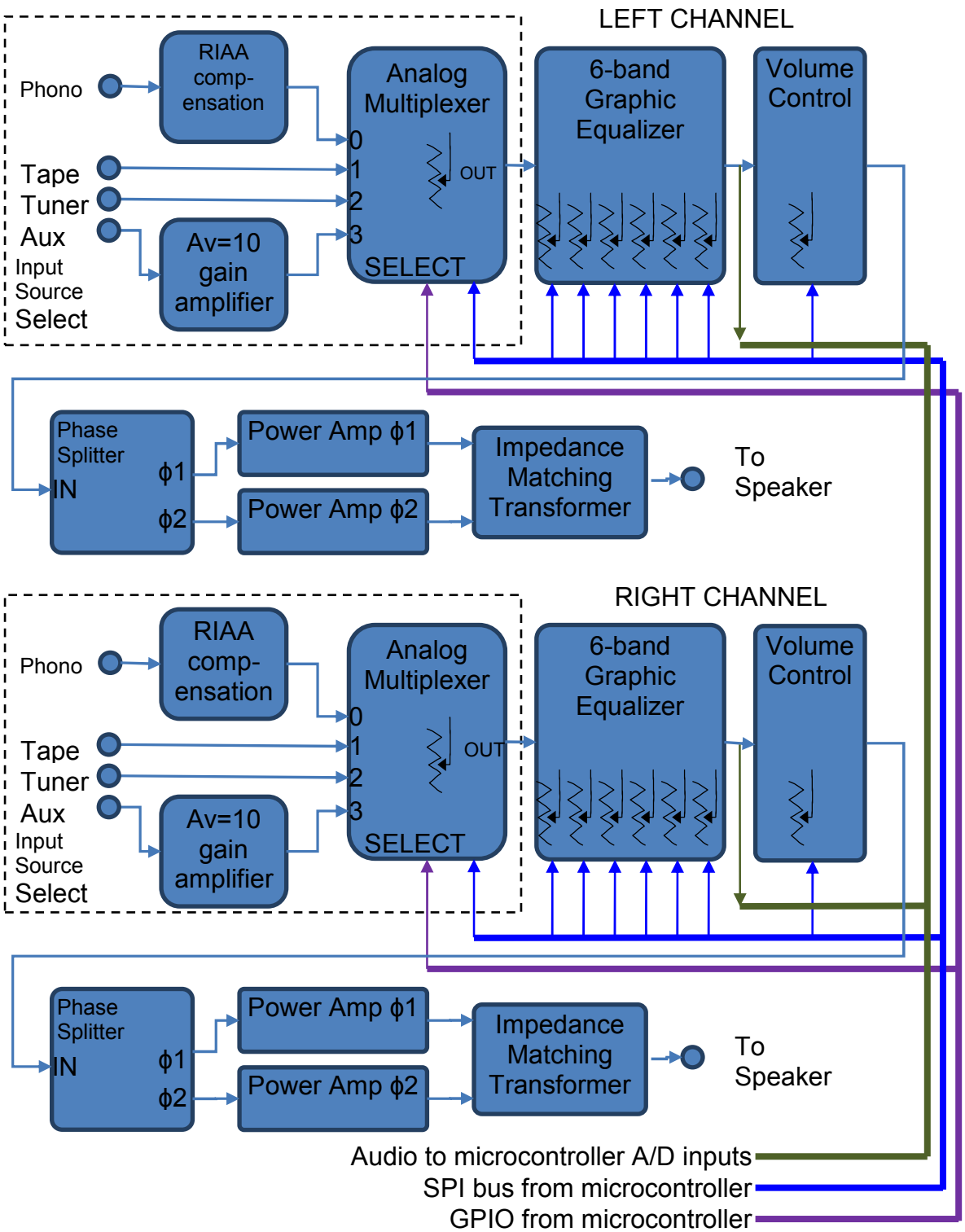


Figure 7: Audio Processor block diagram. (Created by the authors)

4: Project Hardware and Software Design Details

For the project to work correctly and meet the design requirements many subsystems have to work together in harmony. This implies correct interfacing with hardware and software, software and software, and hardware to hardware. Large amounts of attention have been put toward making sure all these systems work together.

4.1: Software Block Diagram

Figure 8 below represents the high-level design of the software modules and how they will interact with each other. On the left side of the diagram the action is triggered by the user; the Touch Screen module receives the input through the XPT2046 Controller and passes the details to the Digital Equalizer. The right side of the diagram received its input from the audio being played; it passes to the Digital Equalizer the values. The digital equalizer indicates to the GUI module whether it shall display a visualization or the Equalizer screen and the needed values. Graphic User Interface (GUI) module generates the next state of the LCD screen and gives it to Display module which then updates the LCD Screen through the SSD1963 Controller. Each of these modules will be broken up into several sub-modules in order to achieve better development.

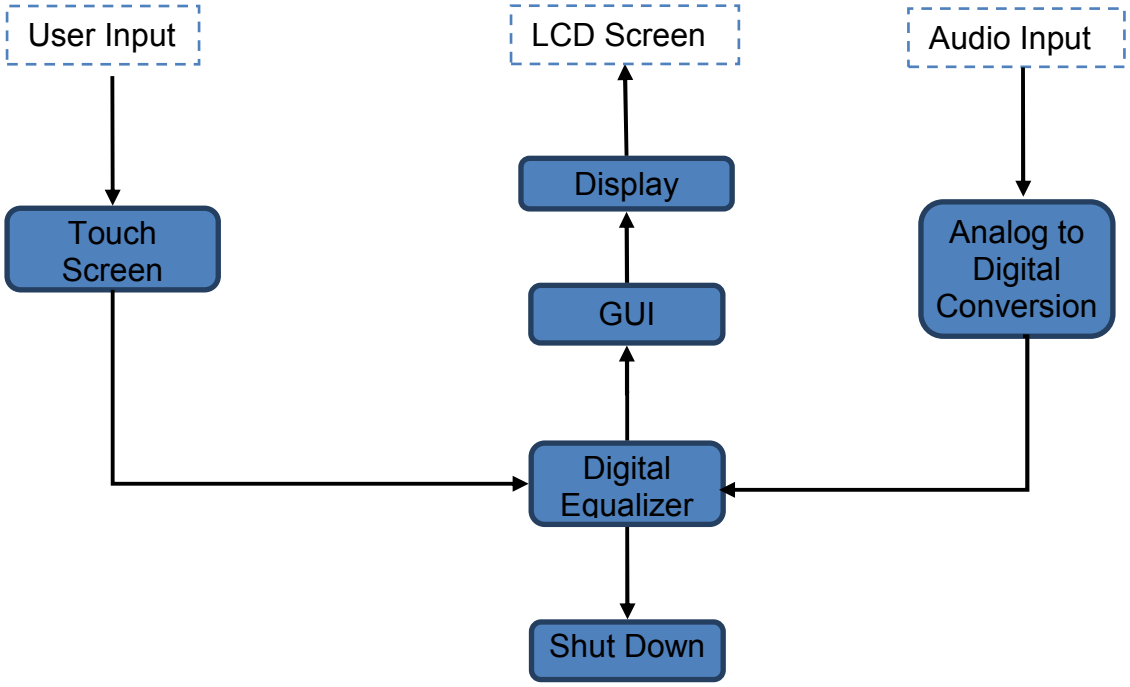
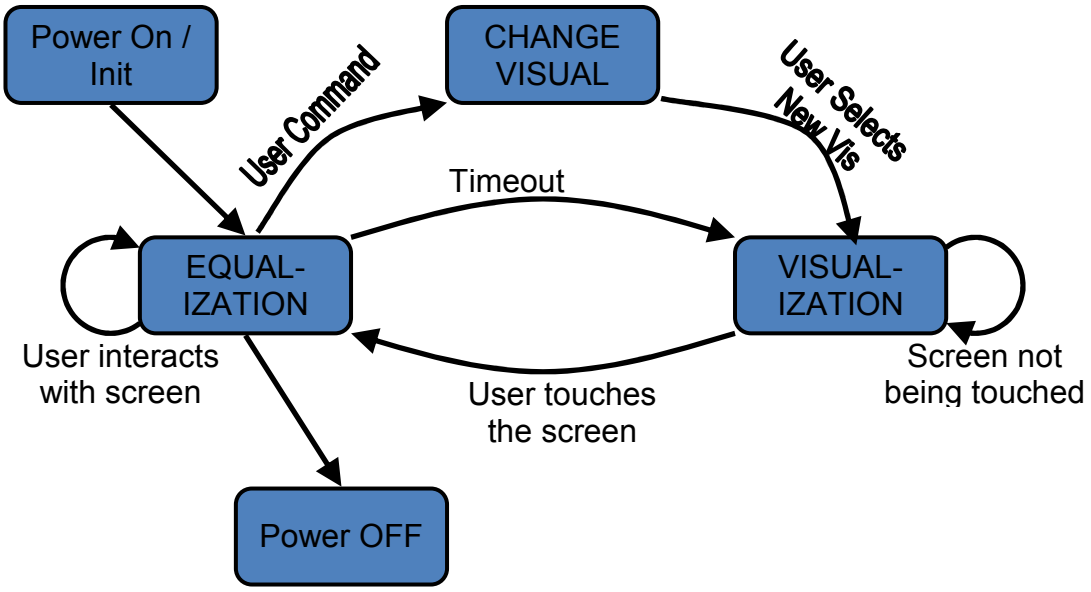


Figure 8 Software Block Diagram

Figure 9 below shows the states that the finished project can be used in.

Figure 9: Software State Diagram. (Created by the authors)



4.2: Physical Construction

Refer to Figure 10 for the following discussion. The finished product consists of multiple CCAs and chassis-mounted components: A1 and A2 are identical printed circuits; each comprises the input source select, audio processing and power amplification for one channel. A3 is a printed circuit comprising the low voltage power supplies, microcontroller, interface to the touchscreen / LCD module, and a bank of optocouplers to isolate the high speed digital circuitry therein from the largely-analog A1 and A2 cards. A4 is vector-board assembly consisting of a few fairly large capacitors, resistors and a choke. A5 is a point-to-point assembly consisting of a rectifier and very large capacitor. The remaining components will be individually mounted within the chassis. See Figure 11 for a photograph of the finished project.

Figure 10: Internal Chassis Layout (created by the authors)

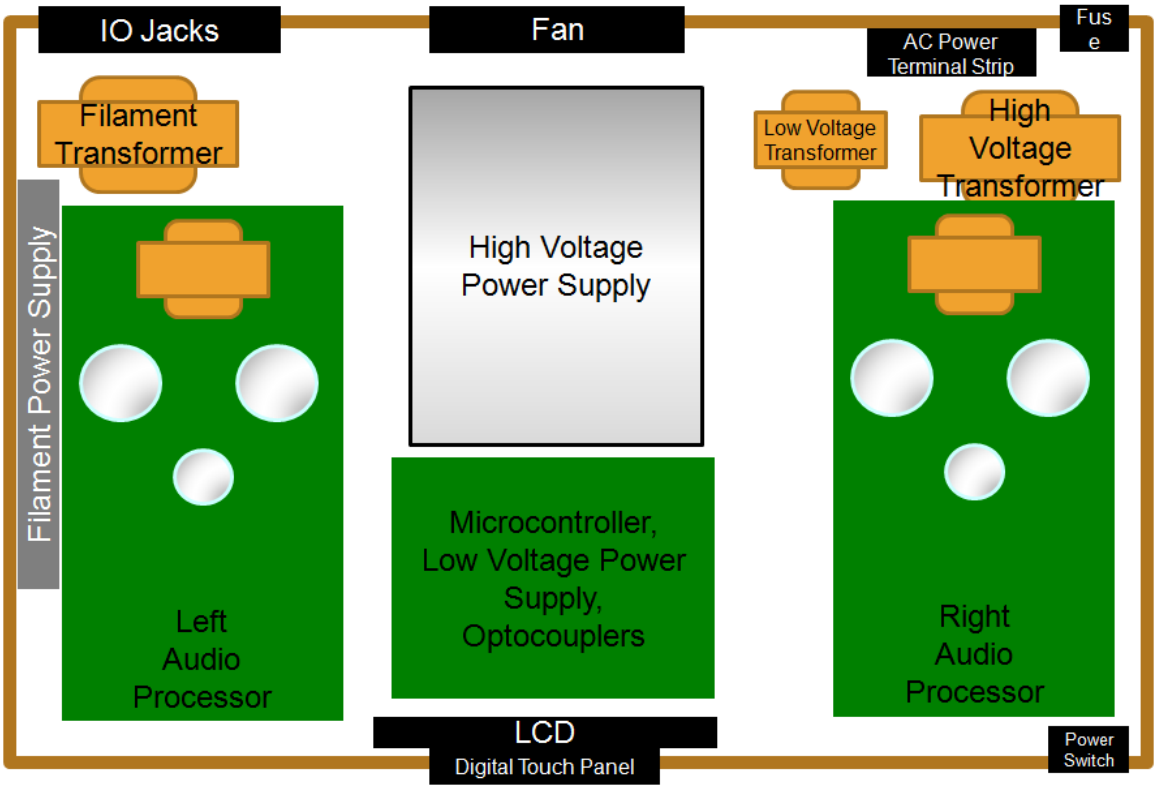
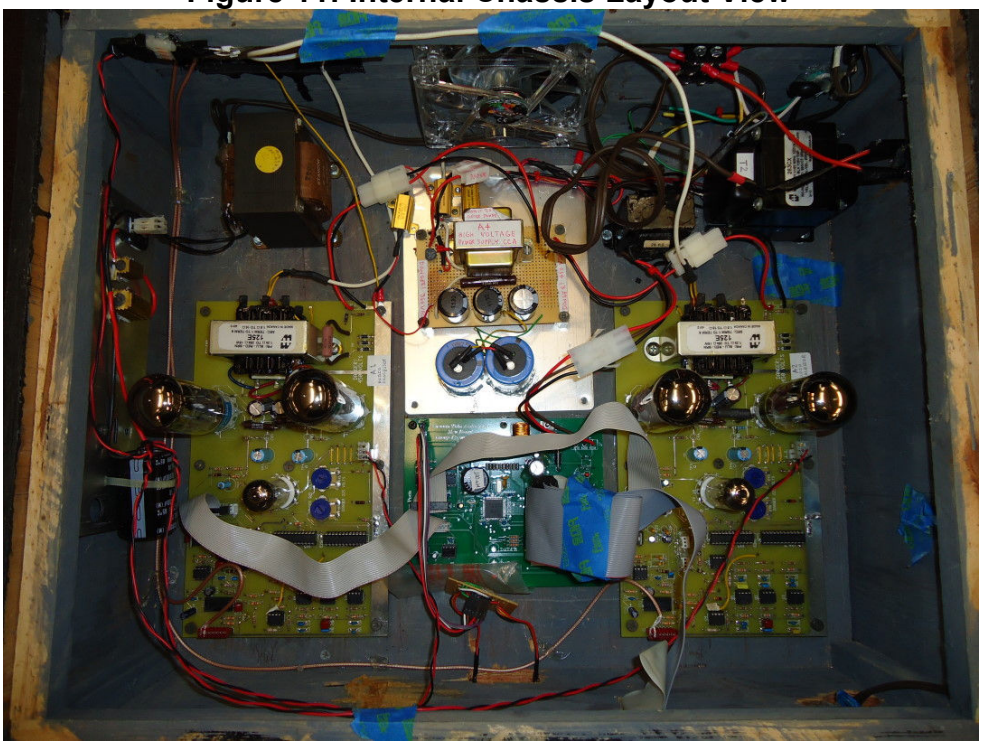


Figure 11: Internal Chassis Layout View



4.3: Input Source Select

Refer to Figure 13 for the following discussion. The finished product is designed to accept up to four audio sources, which enter via commonly available “RCA” jacks. The “Phono” input accepts a signal directly from a magnetic type phonograph cartridge. The signal passes through an operational amplifier filter, derived from a reference design in the LM4562 data sheet, with the complementary filter characteristic to compensate for the Recording Industry Association of America (RIAA) specification applied to vinyl when they are recorded. This curve is shown in Figure 12 [5].

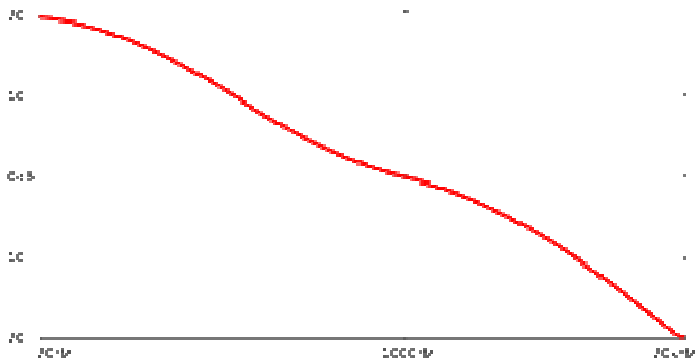


Figure 12: RIAA equalization curve

The “Tape” and “Tuner” inputs are designed to accept “line level” audio of nominally 1V peak value go directly into the multiplexer, each with 10K ohm resistors to ground to provide some Electro-Static Discharge (ESD) protection to the otherwise direct to CMOS input. The “Aux” input has an operational amplifier configured to provide a gain of 10 to enable the use of lower level audio inputs. An analog multiplexer is used, driven by two GPIO from the microcontroller to select one of four input sources. The multiplexer is an 8:1 CMOS analog multiplexer with only the lower four inputs used (the unused upper four inputs are grounded). The digital inputs and all voltages are provided from the microcontroller CCA. The resulting signal is called AUDIO_SEL.

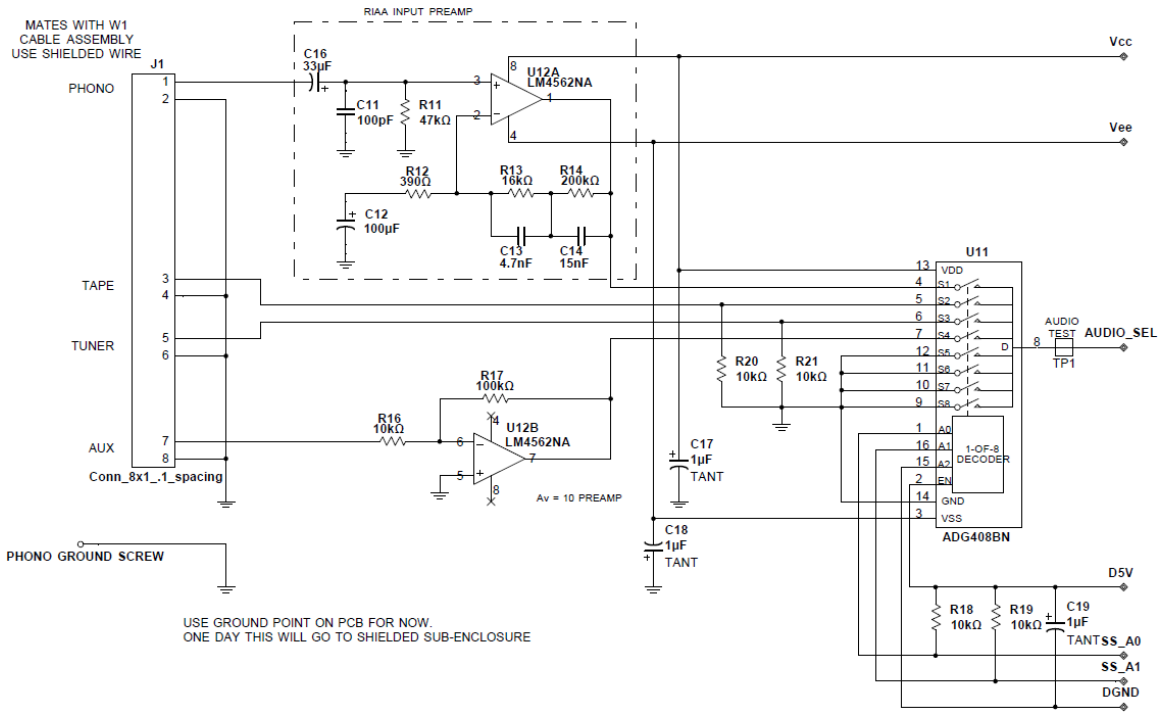


Figure 13: Schematic of Input Source Select circuit

4.4: Audio Processor

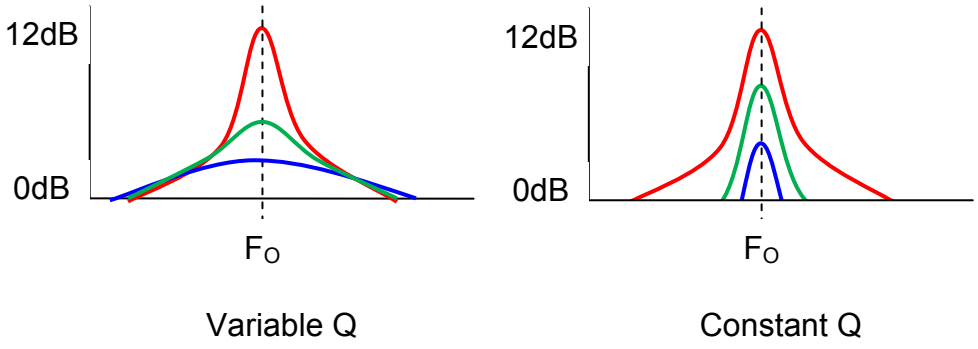
Audio equalization will be performed to adjust the perceived output signal. There were two types of audio equalizers that were considered in the design and the research of the project. To start there are two main classes of equalizers which are active and passive. Active equalizers can't work without an outside power source; they get their name because they are realized by the usage of active components such as op amps. Passive equalizers do not use active components; instead, they employ the usage of inductors to get the needed increase in order. Inductors are needed in passive filters to increase the quality value, but give off and pick up a large amount of electromagnetic (EM) noise. Since these circuits are going to be in close proximity to large transformers and high frequency digital components the amount of resistance to noise needed to be high. Also the introduction of EM noise into the analog amplifier system would cause undesired side effects. Also passive filter topologies are normally larger in scale due to the large inductor size often needed and since PCB place is at a premium this is a problem. Based on the above argument the decision was made to use an active filter design versus a passive filter design for the equalizer circuit. Active filters are advantageous because they can achieve a higher order without the need of introducing inductance to the system via expensive, large inductors.

Within the group of active equalizers there are several classes of equalizers that were considered, namely, graphic, parametric, and non-adjustable. A graphical equalizer gets its name from because the gain for each frequency band is

normally adjusted by a sliding potentiometer. By looking at the position all the sliders in a row a rough idea of the frequency response can be gained in most cases. Also with a graphical equalizer only the gain of the center frequency of a specific frequency band is adjustable, all other values are set with capacitor and resistor values. A parametric equalizer offers more control than the normal graphic equalizer. In a parametric equalizer not only is the gain adjustable but the center frequency, and the bandwidth are also controllable. This gives the user more control over the desired signal manipulation but at a cost. By having so many variables adjustable the complexity of the circuit needed to realize the desired characteristics. Final there is a non-adjustable equalizer and based on the name one can denote its properties. None of the parameters on a non-adjustable equalizer are variable which does not suite the project design at all. So the discussion falls between parametric, and graphical. Due to the complexity of the circuitry needed to make an equalizer with multiple channels that is fully parametric, and the fact that only the amplitude of boost or cut at the desired frequency needed to be adjustable, the graphic equalizer approach was chosen.

There is another means of classification of equalizers, that is, do they have a constant quality value, or non-constant quality value. Based on the information provided by Raine Corp[2], audio control the bandwidth changes with the amount of boost and gain applied to the circuit. The equalizer circuit will only perform at its designed specification at a specific frequency if the quality value is not constant. The quality factor is defined by the dividing the center frequency by the channel bandwidth, so by not having a constant Q the bandwidth will change as the amplitude of boost or cut is changed. Figure 14 below is a graph that displays the difference between having a variable Q value and a constant Q value. As you can see a desired gain at a specific frequency is difficult to reach due to flattening of the frequency response at lower gains.

Figure 14 (Created by the authors)



In order to obtain a constant Q value equalizer there must be another stage added into the design. This is because in order to maintain a constant Q the gain of the filter cannot be adjusted due to the fact that the gain and the quality factor are directly coupled. Now in the design there must be a separate stage to adjust the gain of the pass band. By having this extra stage the gain is now decoupled from the Q factor which gives the desired constant Q functionality along with adjustable gain. Because each band-pass filter has a different center frequency this mean they also have different gains, because in order to have a common Q factor and different center frequencies the gain has to be changed to allow for this to happen. So this means is that the output of each band-pass filter will have a different gain. Well if every band-pass output is feed into the same summing amplifier then the ratios cannot be the same on each phase to get the same amount of gain. If the design were to just implement two stages, one for filtering and then another for adding gain or attenuation, then the same components would not be reusable and also different design would have to be realized for each of the gain stages. To compensate for this a intermediate gain adjuster stage can be added into to normalize the output of each filter to a constant gain then the normalized output could be piped into a large summing amplifier with a bunch of identical components. So with this design idea there would have to be a total of 3 active stages in order to fully process the signal.

Two equalizer topologies were considered, and evaluated on NI Multisim. The first equalizer considered was a constant-Q 6 banded 3-stage equalizer, implementing a second-order bandpass Sallen-Key topology with a Butterworth filter signal response. Figure 15 is the circuit diagram for the 300Hz filter that would be implemented.

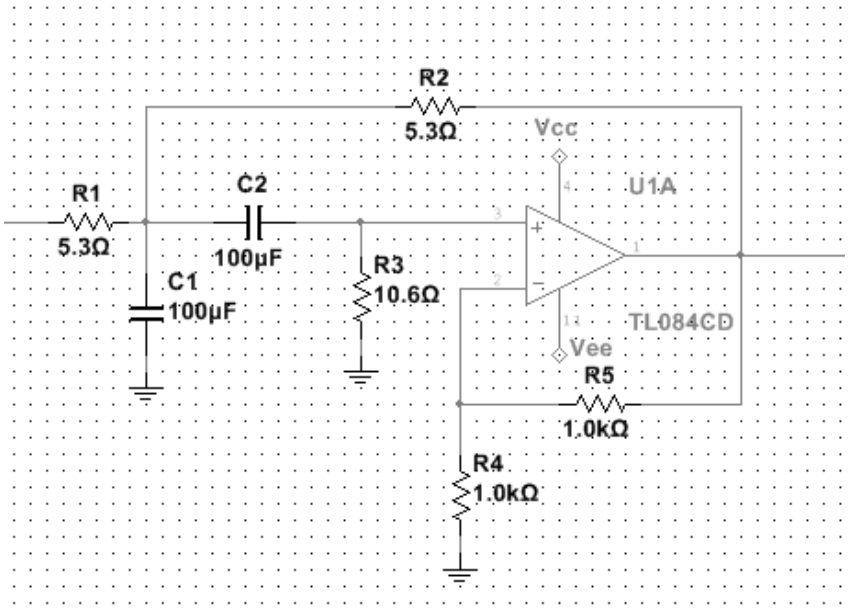


Figure 15

This circuit would have been implemented six times for each channel, with different component values for each band. Since the finished project is stereo, a total of twelve filtering circuits are required for just the equalizer. The next stage (not shown) is the gain correction stage consisting of a voltage divider and a unity gain buffer to normalize the output gain to a constant level of 1.

After the output of each of the filters has been normalized they are recombined in the summation stage. The summation stage is where the gain will be adjustable and where the digital control system would interface with the analog signal path. There will be a digital potentiometer, one per band, in series with the output of the gain normalization stage. By adjusting this digital potentiometer the gain for the specific frequency band can be either boosted or cut. The feedback resistor will be held constant. The summing stage is shown in Figure 16.

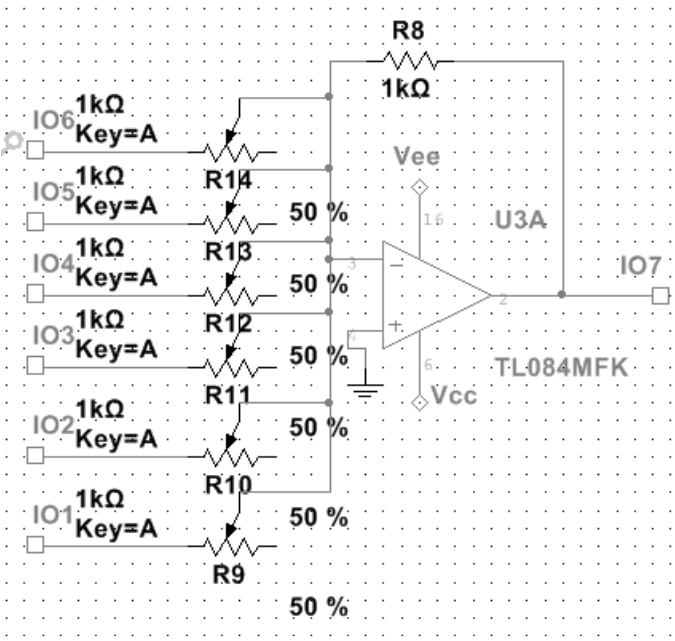


Figure 16

The following equations can be used to calculate all the desired values for the band-filter stage.

$$\omega_0 = \frac{1}{R * C} \quad R = R1 = R2$$

$$\frac{1}{Q} = 3 - K$$

$$K = \frac{R_b}{R_a} + 1$$

The second equalizer topology considered was a gyrator-based design. Refer to Figure 17 for this discussion. The AUDIO_SEL signal, in the upper-left of the

diagram, passes through a potentiometer, called the “Pre Equalization Gain Pot”. This potentiometer is controlled automatically to account for differences in amplitude between the four input sources and shall be set to a level that prevents premature clipping, roughly 1.5 volts. From there, the audio enters the graphic equalizer, based on a reference design in the LM4562 data sheet, which forms a circuit used by some commercial equalizers known as a “gyrator band-pass amplifier”. The six gyrators are each formed by two resistors and a capacitor together with an operational amplifier to simulate an inductor, in series with another capacitor to form a series-resonant circuit “*routed between the amplifier's inputs. When connected to the positive input, it acts as a frequency selective attenuator; and when connected to the negative input, it acts as a frequency selective gain booster”* [2]. To minimize costs and circuit complexity, only six bands are provided. The six frequencies chosen are roughly geometrically spaced and centered within the audio band. The center frequency of each band is determined by:

$$f_c = \frac{1}{2\pi\sqrt{R_a R_b C_a C_b}}$$

Where The $R_aR_bC_a$ terms are the components associated with the gyrator and the C_b term is the remaining capacitor.

Selecting the bands in this manner provides a very cost-effective circuit. Resistors R_a and R_b were made the same for each gyrator, requiring only different C_a and C_b values. Using standard component values provides the center frequencies listed in Table 15.

Table 15: Computation of center frequencies where X corresponds to the reference designator numbering on the schematic.

Gyrator # X	CXa (F)	CXb (F)	RXa (Ω)	RXb (Ω)	Frequency (Hz)
1	100.0E-9	10.0E-6	62000	470	29.5
2	39.0E-9	2.2E-6	62000	470	100.7
3	10.0E-9	1.0E-6	62000	470	294.8
4	3.9E-9	220.0E-9	62000	470	1006.5
5	1.0E-9	100.0E-9	62000	470	2948.3
6	390.0E-12	22.0E-9	62000	470	10065.4

The output of the equalizer splits into two paths: One path is via the user adjustable volume control, resulting in a signal called AUDIO_EQ that passes to the tube amplifier and the other to the a level shifter and bias circuit that insures that the 1.5 volt maximum level is attenuated and centered within the optimal range expected by the A/D input on the microcontroller, resulting in a signal called is called AUDIO_SAMPLE.

See Figure 17 for a schematic of the analog portion of the graphic equalizer.

Refer to Figure 17 and Figure 18 for the following discussion. All potentiometers in the audio processor are digitally controlled, which is a unique feature of this project compared to commercial products. The AD8403 quad digital potentiometer is used, which is controlled by an SPI-bus-like signal from the microcontroller. (See paragraph 4.13)

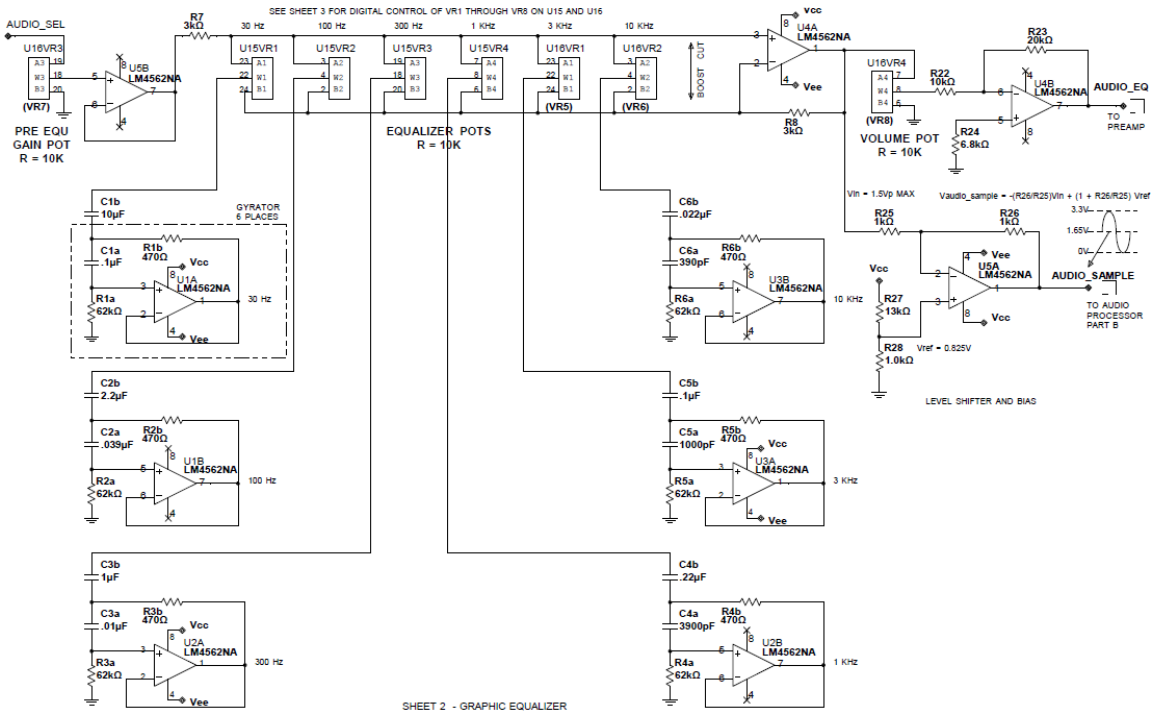


Figure 17: Analog portion of the graphic equalizer.

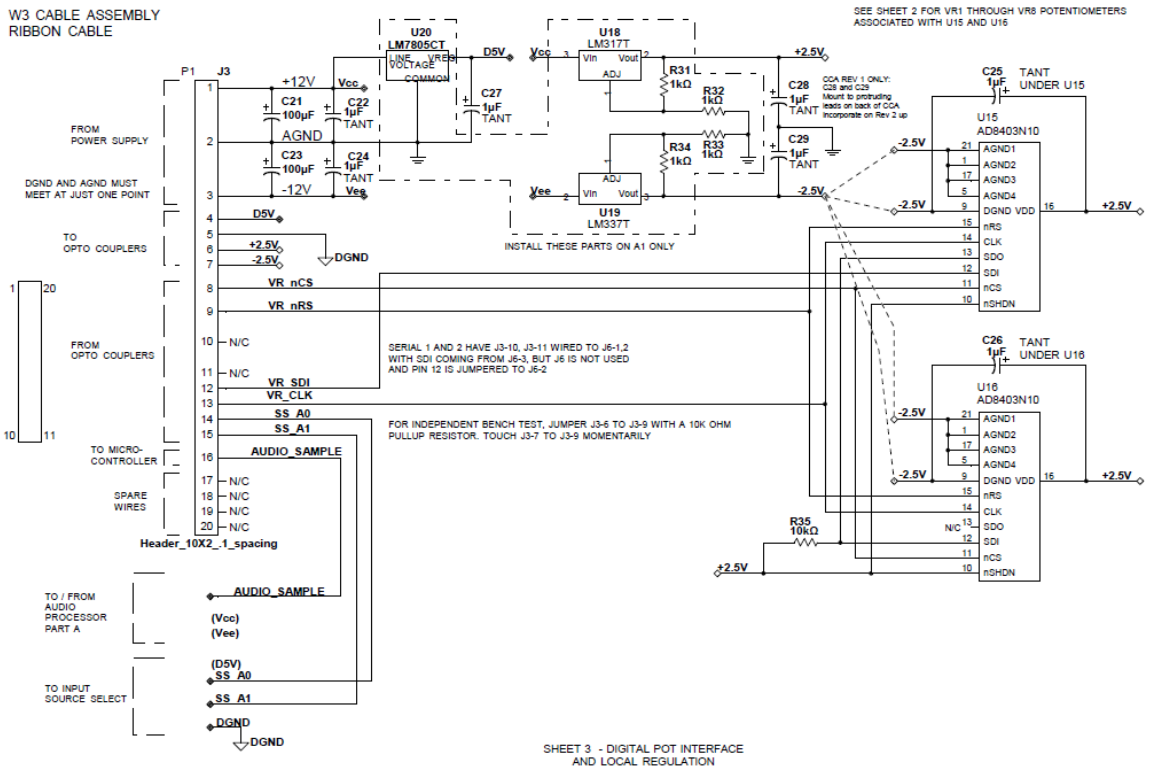


Figure 18: Local voltage regulation (top) and digital portion (bottom and right) of the graphic equalizer

4.5: Vacuum Tube Audio Preamplifier

Refer to the left side of Figure 19 for this discussion. The phase splitter resulting from the reference design produced in the trade study proved to be a source of excessive distortion in the early simulations, as well as being overly complex. It was replaced by a simpler design using a differential amplifier similar to the input stage of a MOSFET-based op-amp studied in UCF's Electronics II class. The AUDIO_EQ signal is biased to +2.0 volts, a value determined by experimentation that provides the lowest distortion. The lowest distortion output from the final power stage is achieved with slightly unequal amplitude signal voltages and very specific phase splitter plate voltages. To facilitate this, both of the phase splitter's plate resistors are both replaced with mechanical potentiometers to allow adjustment. The final value will be determined during testing. The phase splitter requires a lower supply voltage that the power amplifier stage, however current draw is just a few milliamps for each phase. A stacked zener diode series provides the regulation.

4.6: Vacuum Tube Power Amplifier

Refer to the right side if Figure 19 for this discussion. The 2.95% THD offered by the reference design produced in the trade study circuit leaves a lot of room for improvement. Other push pull amplifier configurations were designed and

simulated, however none worked as well or even provided substantially better performance. Given that scenario the reference circuit was slowly refined. The first improvement was to provide a separate cathode resistor and bypass capacitor for each 6L6. This cut the distortion to around 1.6% THD.

Next, attention was paid to the power dissipation of U1 and U2; Separate measurements not shown revealed a dissipation of 100 watts per tube, greatly exceeding the 6L6 tubes power rating of 30 watts. Many hours of study, calculation, and simulation experiments were used to arrive at the final circuit design, as there were several tradeoffs that had to be made:

1. Maximum output power occurs with the highest plate voltage. High plate voltages unfortunately run the risk of exceeding the 6L6 maximum plate voltage rating of 450 volts.
2. Lowest distortion was achieved with the highest plate voltages.
3. Lowest distortion was achieved with lower values of cathode resistor, however this resulted in higher plate voltages.

The next step was to find the ideal impedance presented to the plate by the transformer / speaker combination. The chosen part number of impedance matching transformer provides six taps on the secondary; various combinations of taps can produce impedances ranging from 3000 to 22.5 KΩ. Impedances on the low end result in high distortion, while those at the high end cause premature clipping and low power output. A value of 5600 ohms was ultimately chosen, resulting in the best combination of maximum output power and THD.

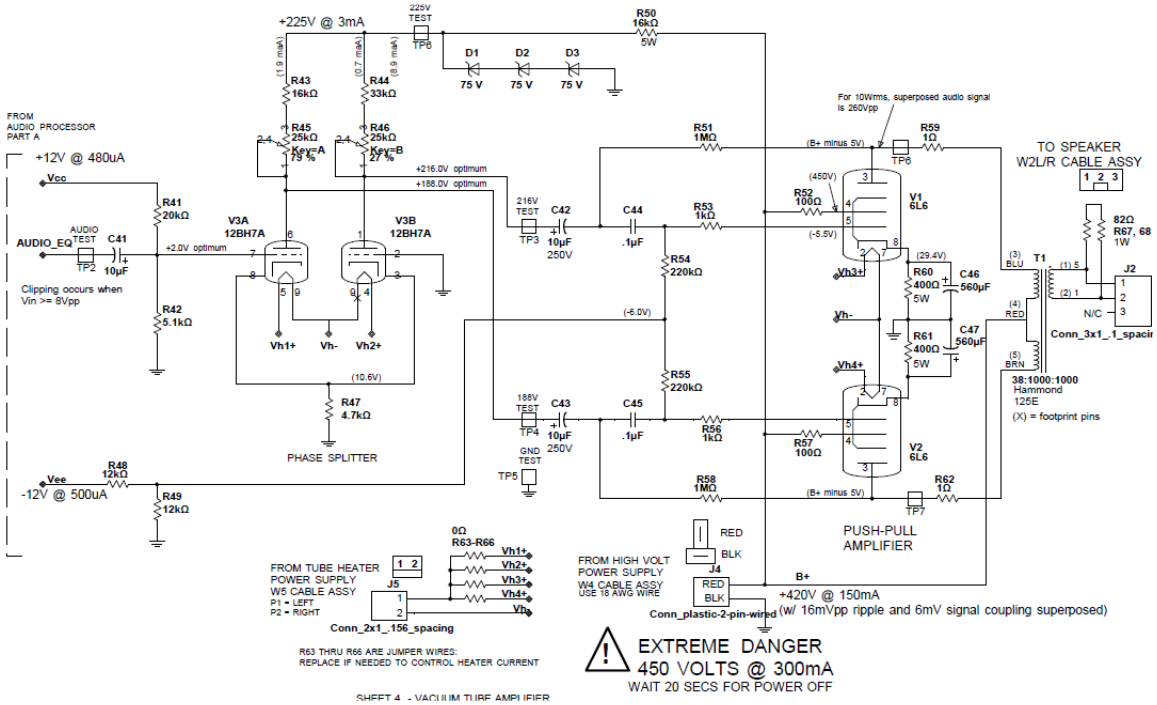


Figure 19

During further attempts to optimize this design, a load line graph was created; See Figure 20. On this graph the blue curves represent the Vg=0 and Vg=12 grid bias voltage curves taken from the JJ Electronic vendor data sheet for the 6L6. The red curve represents the maximum allowable plate power dissipation of 30 watts. The five plotted load lines were measured during several optimization attempts (the diamond symbol on some lines represents the amplifier's quiescent point), and are summarized on Table 16[11] Note that "B+" means the supply voltage for the vacuum tube and R_K is the cathode resistor circuits (analogous to V_{CC} and R_E in a solid state circuit).

Table 16: Results of several optimization attempts.

Line	Condition	Why not chosen
	The Design 1 configuration from the trade study. B+ =470 volts, R_k=180Ω per tube	Exceeds the plate power dissipation rating
	B+ lowered to 260 volts	Did not meet power requirements
	Use a calculated value of B+ =450 volts, R_k=300Ω, common to both tubes	The actual load line doesn't match the calculated one. See next entry.
	The actual load line, resulting from the above values	Simulation reveals excessive distortion: 8.3% THD
	B+ =450 volts, R_k=400Ω per tube	This is optimum

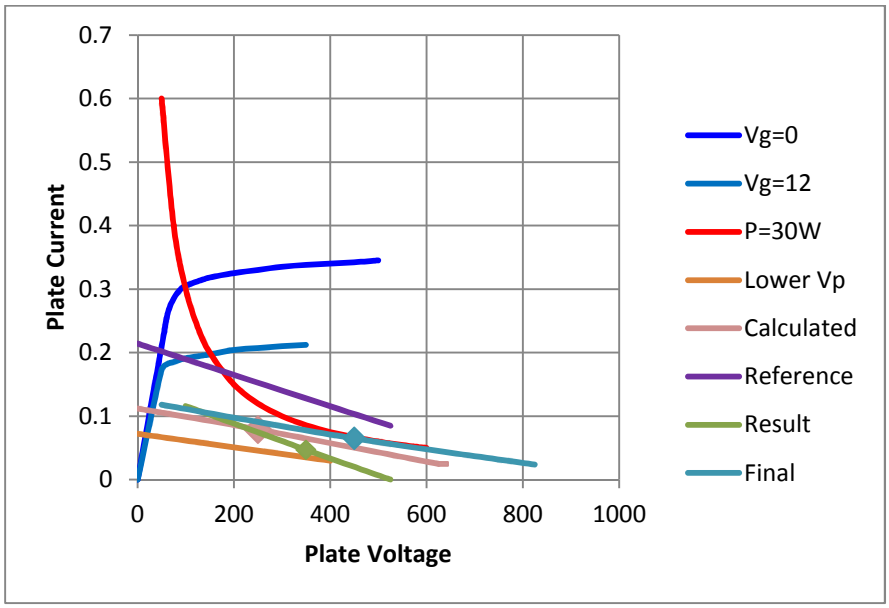


Figure 20: Power amplifier load line using a 6L6 vacuum tube (data taken from one side only)

The resulting load line, which was measured from a simulation of the final configuration of the amplifier, is close to ideal. The line is nearly tangent to the 30W power curve, with a nearly-centered Q point, and gives the lowest distortion

/ highest power characteristic possible in this design, while still meeting the output power requirement. Simulation results of the final design configuration yields the measurements listed in Table 17 with a 1 KHz input at an amplitude such that the output is just under the onset of clipping.

Vin, peak to peak	3.52V
Vout, RMS	9.7 volts
Power output into 8Ω load	11.7 watts
THD	2.1%

Table 17

4.7: Touch Screen

The touch screen been used in this project is integrated with the LCD and its controller. The touch screen controller has the ability to produce an output to the MCU when a user touches the screen or drags a finger across it, and generates a serial output message that indicates the exact coordinates of the touch and how much pressure was applied. This message is used by the software on the microcontroller to determine which action user is attempting.

Using the coordinate values obtained from the touch screen controller, the software will interrupt the visualization and display the Equalizer screen. Once in the Equalizer screen, the software determines which action the user is attempting. Valid actions, such as select a different source, adjust the equalizer settings, etc. can be seen in Figure 5. The touch screen controller built into the LCD panel is the XPT2046**.** This controller is capable of generating its output in 15 clock cycles. The documentation advises that most microcontrollers are not capable of this speed in serial communication. The touch screen is interfaced via only five signals: T_CLK, T_CS, T_DIN_T_DO and T_IRQ.

4.8: Microcontroller

As the central brains of the project it is very important as it must interface with a multiple of systems. Compared to the Analog Processor CCA, the Microcontroller is a simple design but required high-density PCB traces. Since none of the team members were proficient in designing such a board, a reference design was used as guidance on how to layout the microcontroller section. A full schematic diagram was obtained from the manufacturer of the development board (STMicroelectonics, 2013), [25]. Figure 21 shows the relevant portion of this schematic used as guidance.

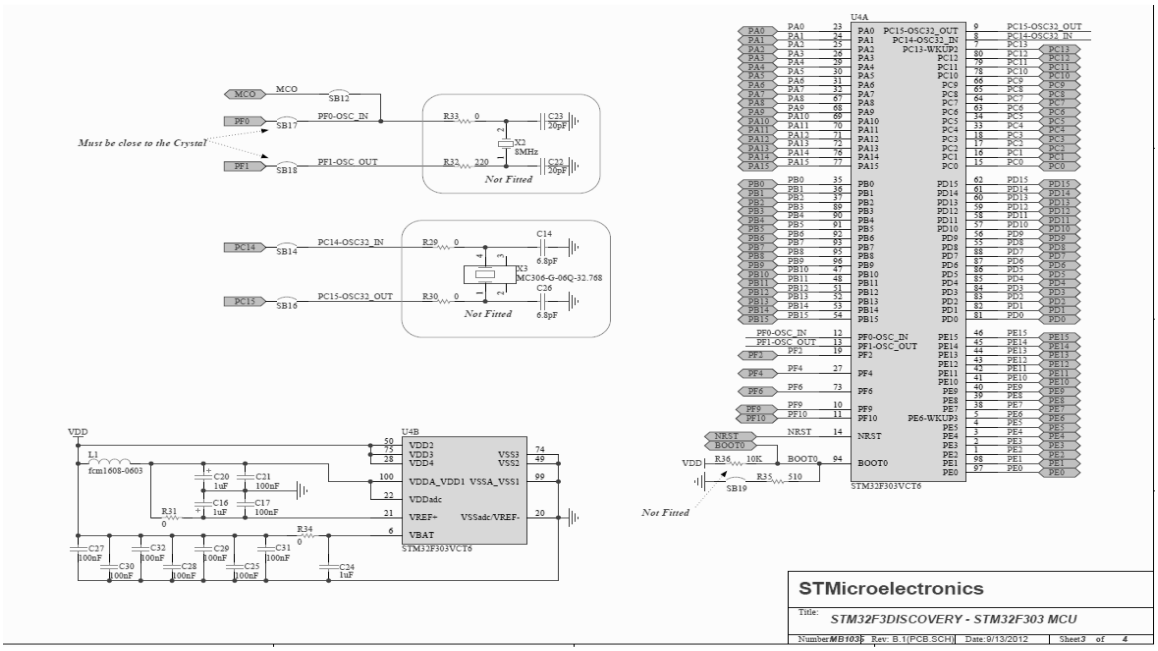


Figure 21

Our design is shown in the following three figures. Figure 22 shows the interface to the programmer J2 in the upper-left, the digital touch interface J1 in the upper-right, the LCD / touchscreen interface J4 at the right, and the clock in the lower left. Note that the touchscreen is supported by this design, but the touchscreen itself is not used (See section 5.11 for details).

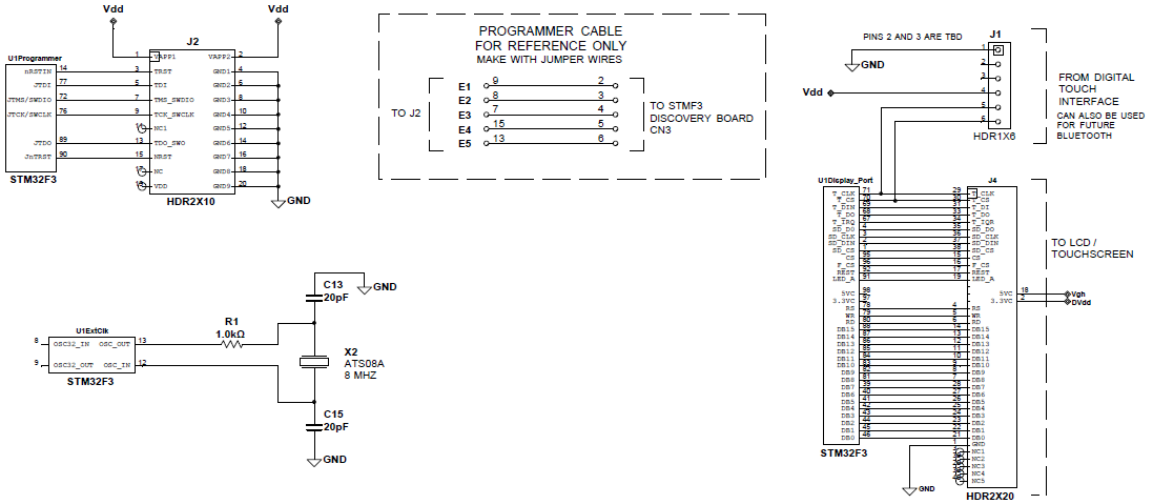


Figure 22

Figure 23 shows the power supply. Most of the voltage regulation is located here because it simplifies power distribution. The regulators supply power as follows: U3 and U6 supply ±12 volts to both Audio Processor CCAs, U4 supplies 3.3 volts to the LCD panel backlight, while U9 supplies 3.3 volts to power the LCD controller. U9 supplies power to the microcontroller, with the analog reference

receiving a pi-filtered version of U9's output to insure noise-free analog to digital conversion. See paragraph 4.10.1 for a complete discussion of the voltage regulation.

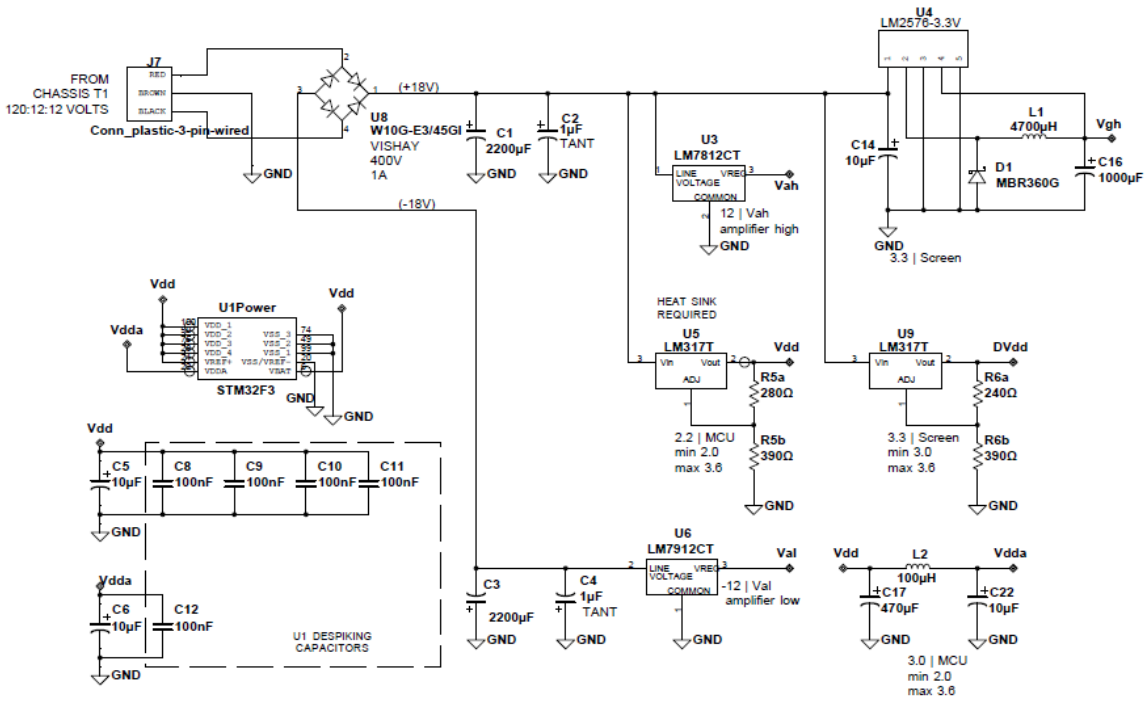


Figure 23

Figure 24 shows the remaining GPIO interface of the MCU and the optical couplers which interface the +3.3V / 0V logic output of the MCU to the ±2.5V logic signals required by the digital potentiometers on the Analog Processor CCAs. Connectors J3L and J3R interface to the respecting CCA via ribbon cables. Note that virtually all signals are in parallel, except for a separate VR_SDI that provides individual serial streams for each Audio Processor.

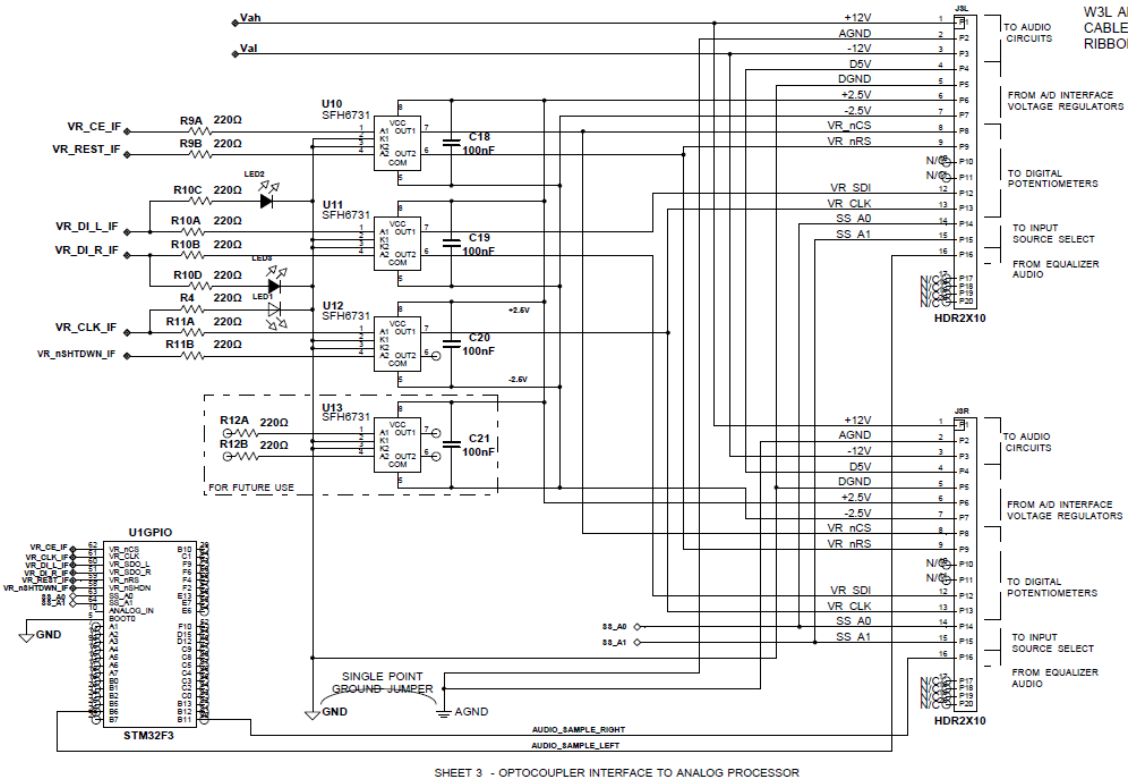


Figure 24

4.8.1: Required Hardware Support

In order for the microcontroller to function properly all the support requirements must be met. During the design and prototyping stage the development board can be used to fully support the microcontroller. The development board has all the necessary hardware components to program and operates all the functions of the microcontroller.

4.8.2: Programming Methods

With regards to programming the microcontroller there are several factors to consider such as what integrated development environment will be used for code development, and what method will be used to transfer the code from the computer to the microcontroller. Table 18 compares the IDEs that were considered for development.

Table 18

	Atollic True Studio Lite	IAR Embedded Workbench	Keil Tools	GCC -ARM
Cost	Free	Free	$250	Free
Code size limited	32 KB	No time limit	No	No
Well supported (large community backing)	Yes	Moderate	Yes	Huge community
Ease of use	Simple to set up an configure	unknown	From what it seems, yes	Very difficult
Debugging capability	Basic debugging features	No power debugging support	Full debugging capabilities	Basic debugging features
Runtime libraries included	Yes	No	Yes	No
User interface	Eclipse plug-in	Eclipse plug-in	Custom	Command line terminal

Clearly, Atollic True Studio Lite is an excellent choice. Based on the fact that the project's code size is anticipated to be <32KB, the code size limit should not be a problem. This IDE supports the hardware debugger what will be used and discussed in the following sections. Another great quality that True Studios has is that it is based on the super popular free IDE Eclipse, which the two CE team members are very familiar with. Another huge consideration was the price of the IDE, because this project is self-funded.

The next topic to consider in regards to programming the microcontroller is the method of transporting the compiled binary file from the computer to the microcontroller. The microcontroller that has been picked is capable of being programmed through a standard JTAG 5 wire interface or by a single wire debug SWD interface. Both of these interfaces allow for line by line code debugging, execution tell breakpoint, real time register viewing, and memory viewing. By having this capability it will be possible to debug code once it is installed onto the custom PCB.

All software will be written in the C programming language on a Windows-based host computer, then "flashed" to the target MCU CCA using the development board and appropriate cabling (the discrete-wired portion of which is shown at the top of Figure 22).

4.8.3: STM32F3Discovery

In the project hardware specific code must be developed. In order to be able to develop code while the hardware is still in the design, and prototyping phases a development board is needed. ST provides a vast array of development and evaluation boards for their microcontrollers. The development board that is provided for the STM32F3 series is called the STM32F3Discovery. The STM32F3Discovery features the STM32F303VCT6 microcontroller (same as is used on the MCU CCA), 256 KB Flash, and 48 KB RAM in a LQFP100 package. The STM32F3Discovery includes everything needed to start developing coded. Table 19 lists the features which are very useful for this project. [25]

Table 19

Features	
Programmer	On-board ST-LINK/V2 with selection mode switch to use the kit as a standalone
	ST-LINK/V2 (with SWD connector for programming and debugging)
	Board power supply: through USB bus or from an external 3 V or 5 V supply voltage
Power supply	External application power supply: 3 V and 5 V
GPIO	Extension header for all LQFP100 I/Os for quick connection to prototyping board and easy probing
User Inputs	Two pushbuttons (user and reset)

4.9 Color LCD display

The Color LCD Display (see figure 24) will serve as the user interface. The LCD Screen will display to the user all the information necessary to utilize the system and configure the system. The software will use the screen to display visualizations while music is playing, which react to the audio frequency, amplitude, and phase characteristics in a manner similar to Microsoft Media Center. Visualizations generated will cycle randomly, and last several minutes each. Figure 25 provides the dimensions of the panel, which is about .25” thick

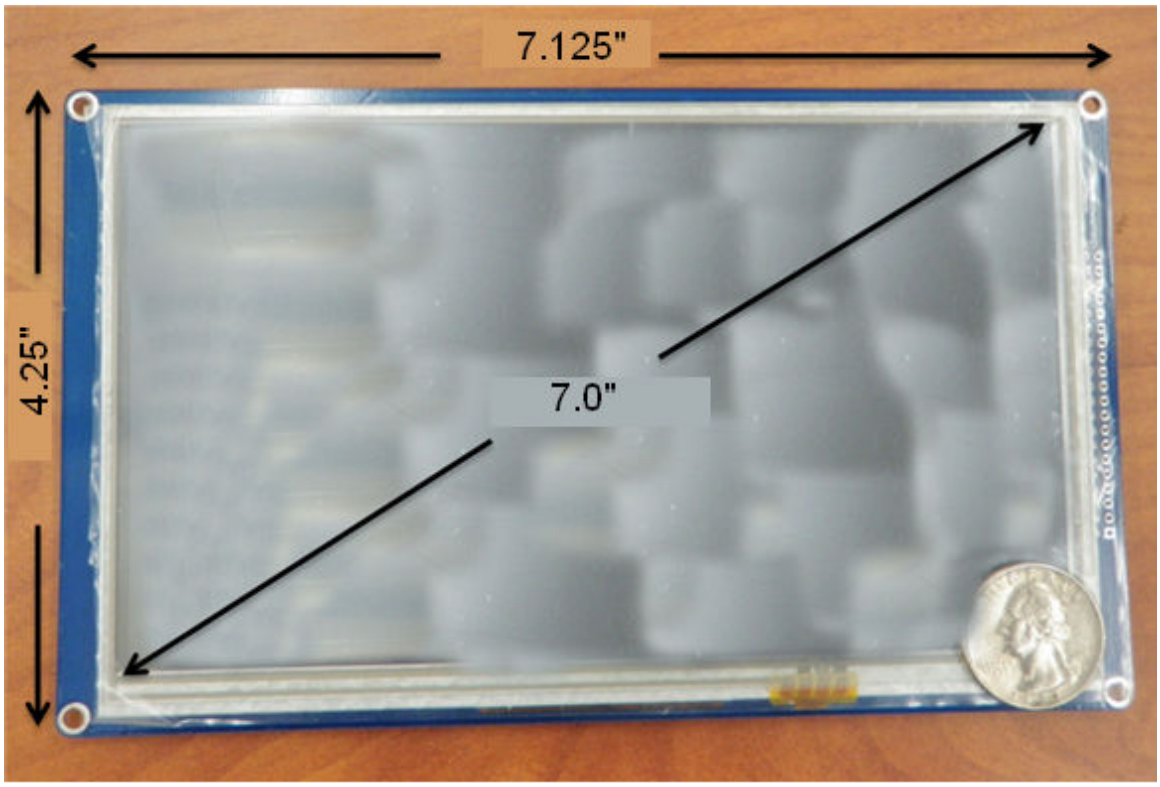


Figure 25 LCD panel dimensions

This LCD consumes approximately 300mA at 3.3V, with 3.3V being provided to both the controller circuitry and the backlight. It will be used to display the splash screen and visualizations shown in Figure 4 and the control screen shown in Figure 5.

The visualizations will be displayed in real time, and it has been determined that to achieve this, a minimum speed of 50ms update time is needed. Therefore, the LCD will be refreshed every 50ms or faster. This LCD screen is capable of handling a refresh time of 8ns, so achieving real time screen update lies in the effectiveness of the algorithm and controller.

The controller integrated in the LCD panel is the SSD196 which is specially designed to control graphic material. It has its own internal buffer and supports TFT 24-bit color interface. The software on the MCU will communicate to the LCD over an 18-bit bus, with each 18-bit “word” representing a single pixel’s RGB color. The MCU will send the data for each pixel and drive the control signals to the SSD1963 in order to synchronize the LCD pixels in the right order.

The SSD1963 controller has the ability to rotate the graphics 0, 90, 180, 270 degrees; however this feature is not required. The LCD panel contains an SD card reader; however it was not used for this project.

This LCD / Touchscreen proved very troublesome to interface and required a great deal of time to program correctly see paragraph 5.15 for the difficulties that were encountered during development.

4.10: Power Supply

It would have been preferable to develop a single supply for the entire project, but the requirements were simply too different to be feasible, thus there are three power supplies required: A low-voltage / low current power supply for the digital components and some of the low power analog components, and a high-voltage / low current power supply to provide power to the vacuum tube plate circuits, and a low-voltage / high current power supply for the vacuum tube heaters.

Refer to Figure 6 for this discussion. The finished product is designed to operate using 120V / 60Hz utility power. A three-prong grounded power cord of the type used for desktop computers supplies power to the unit via an IEC320-C14 type chassis-mounted connector J7. The ground lead will terminate at a chassis ground point and have continuity to all exposed metal parts (via a dedicated ground wire if necessary) for user safety. The two 120 volt leads will terminate at a two-position screw terminal barrier strip. The barrier strip will be equipped with a plastic cover plate for safety during test and servicing. Power from J7 shall be fused. 120 volt power is supplied to the transformers of the various supplies.

4.10.1: Low voltage

The low voltage power supply will provide power to all the digital components of the system as well as the operational amplifiers on the Analog Processor CCA. Table 20 summarizes the power requirements for this supply.

Table 20

Circuits powered	Symbol	Voltage	Approx current
LCD Backlight	Vgh	+3.3 V	200 mA
LCD Controller	DVdd	+3.3 V	100 mA
Analog Processor +	Vah	+12 V	100 mA
Analog Processor +	Val	-12 V	100 mA
MCU digital circuits	Vdd	+3.0 V	160 mA
MCU analog reference	Vddref	+3.3 V	<1 mA

The low voltage power supply consists of a center-tapped transformer providing 26Vpp. The output of the transformer will be passed through bridge rectifier, filtered and submitted to a bank of regulators, each representing a voltage output from the requirements. Linear regulation is used throughout that U4 is a switching regulator to avoid due high power dissipation. See Figure 23.

4.10.2: High voltage

Refer to Figure 26 for the following discussion. 360 volts AC from T2 is supplied to the High Voltage Power Supply CCA, A4, which is responsible for generating the 450 volts DC at 300 milliamps required to operate the vacuum tube plate circuits. This supply is a very basic DC supply with a PI-type filter. A more complex IC-regulated design was investigated, but deemed unnecessary since vacuum tube amplifiers made in the 20th century generally lacked regulators, or used very simple regulators based on vacuum tubes. This power supply is unregulated, but provides very low ripple. Output voltage is approximately 490 volts unloaded or upon power up, but lowers to approximately 400 to 420 volts as the tubes warm up and craw current. P1 and P2 supply power to both Audio Processor. **DANGER: The High Voltage Power Supply CCA, A4 generates extremely dangerous, if not lethal, voltage.** For this reason, A4 was the design and test responsibility of the most senior member of the design team.

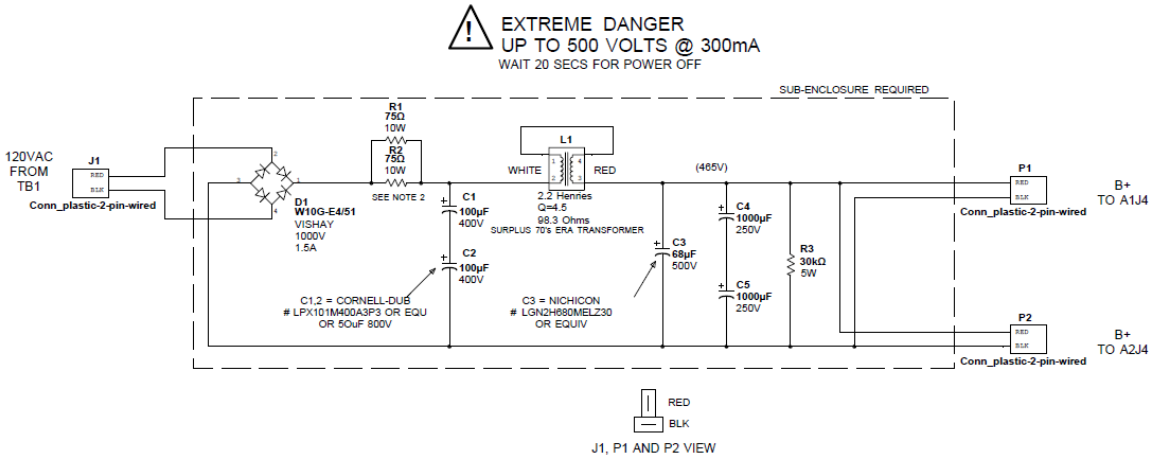


Figure 26

4.10.3: High current

The heaters draw 2.4 Amps at 5.5 to 6.0 volts per Audio Processor, or 4.8 Amps total. This posed a particular design challenge: How to generate this voltage without also generating a lot of waste heat. Older tube circuits simply used AC voltage, but this was rejected due to the potential for hum generation. A number of ready-made supplies were investigated and rejected due to their high cost. During design of a regulated supply, it became apparent that a fairly substantial regulator circuit with a relatively high unregulated voltage would be required, and result in a lot of wasted power. The T2 transformer has an output specifically for vacuum tube heaters, unfortunately it does not have the current capacity.

Ultimately, a transformer was located for free with multiple primaries and secondaries, that when connected in series provided just slightly too much voltage. A simple RC filter, with C very large are all that was required to generate a fairly low ripple 5.75 volts for the heaters. See Figure 27.

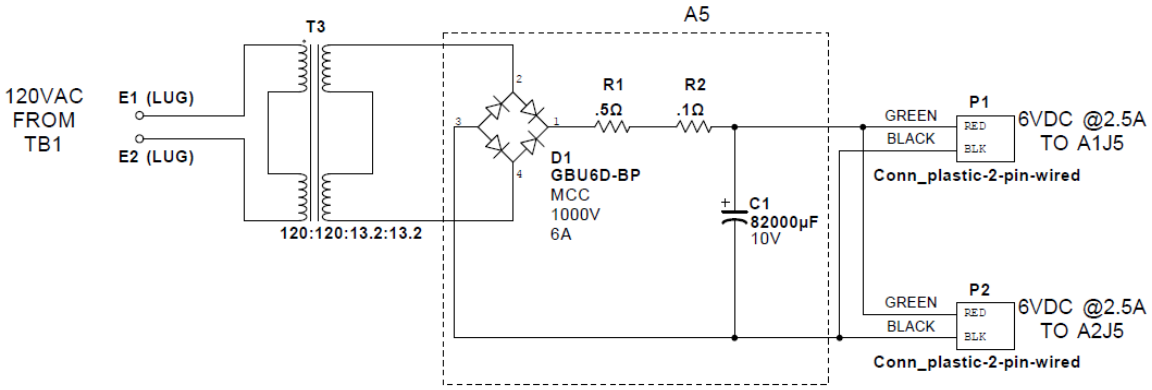


Figure 27

4.11: Power ON Sequencer

(Deleted)

4.12: Graphic User Interface

Figure 28 represents the Graphic User Interface (GUI) of the system, from the Digital Equalizer (input) to the Display module (output).

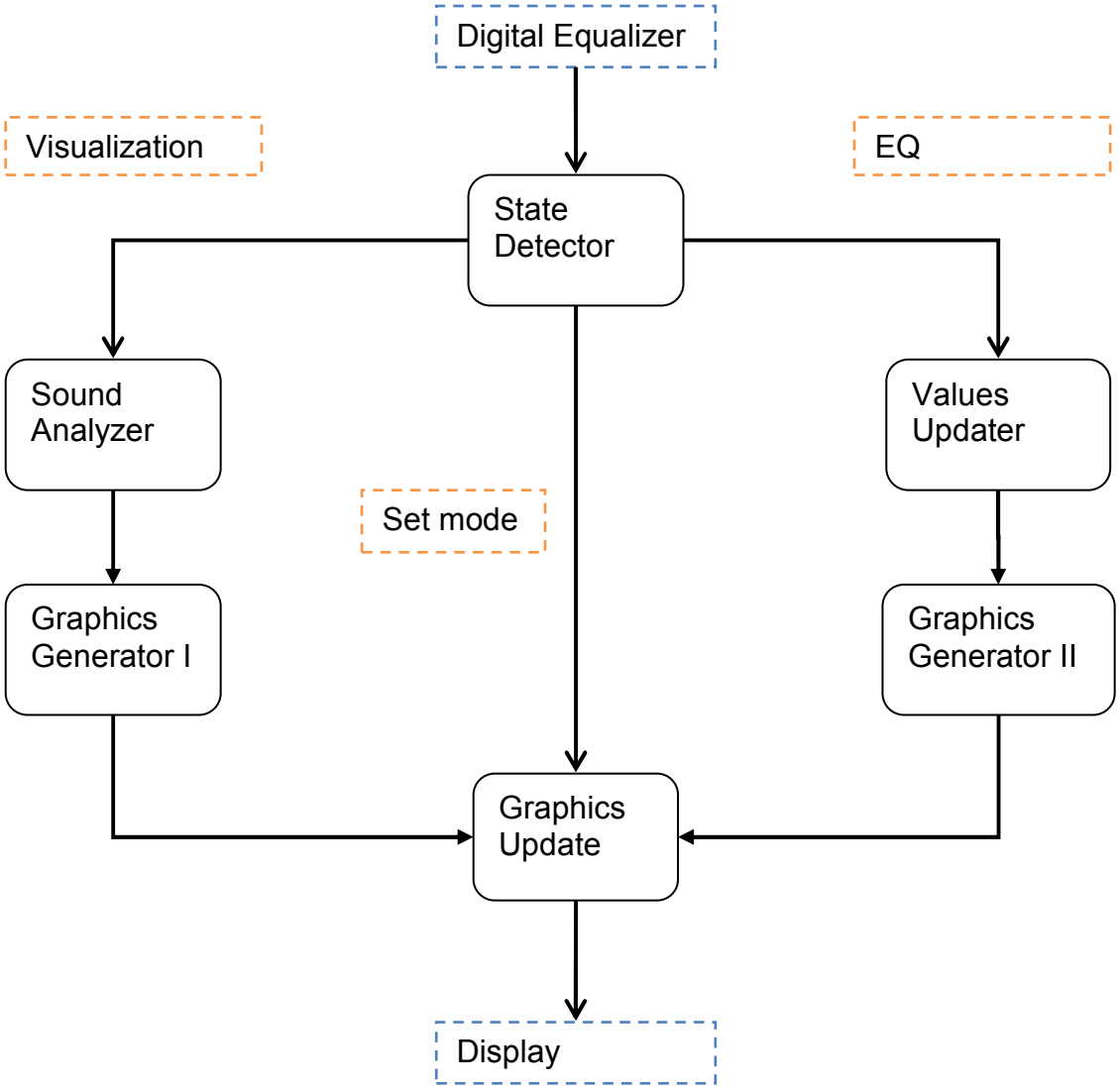


Figure 28 Graphic User Interface System Diagram (Software)

4.12.1: Requirements

The Graphic User Interface (GUI) is the key for the usage of the system. A user depends fully in the GUI in order to understand and communicate with the system. The way the system communicates to the user what its state is, is via the GUI in two ways. The GUI demonstrates to the user that there is music playing by showing visualizations, and it communicates to the user the current configuration and active settings via the equalizer screen.

The GUI is for the user to be able to completely understand what is occurring with the system at all times. Visualization must always be active if the user is not in the process of configuring the system or changing any settings. This will ensure the user knows that there is some input into the system, even if system is

muted. There will not be any buttons or any indication of how to switch into the equalizer screen, however, touching the screen at any coordinates, will activate the equalizer screen.

The GUI lets the user know what all current configurations are via the Equalizer (EQ) screen. This screen must be intuitive. The intention is for the user to look at this screen and be able to understand how to manipulate all the settings of the system without having to possess any computer or technology skills. The input source select, equalization presets, and all the settings shall be easy to find and must communicate current state as well as make user feel comfortable while operating the system.

Aside from selecting the input source, and choosing an equalizer preset, the user must have the ability to adjust speaker balance (left and right); this will be displayed with an intuitive horizontal bar with a marker that indicates current state, user will be able to drag this marker left or right, and the system will adjust volume for the two channels accordingly.

The EQ screen shall also give user the ability to mute the system, the EQ screen communicates this option to the user via an on-screen button. Where the button image has two states, one for when the system is mute, and one for when it is not. The images shall make it obvious that the system is mute or not. Muting the system drops the volume on both output channels to zero (0). In addition, bringing the system back from mute will restore the volume value of both channels which were active before system went into mute.

The user will have the ability to switch back to visualization via an on-screen button which must display text or symbol that makes it intuitive and obvious that pressing this button will close the EQ screen and go back to the visualization screen. Although this button will be available, it is not necessary, for the system will automatically switch back to visualization after it detects inactivity for several seconds. The user must have an on-screen button to turn OFF the display. The screen will “wake up” next time user touches the screen. When the system is woken up, it will take user to the EQ screen.

The EQ screen will offer the user the ability to set different amplification values to six (6) different frequencies. Each frequency will be a vertical bar with a marker indicating current value, and user will have the ability to move this marker up or down in order to set a new value to a frequency. There will be preset equalization values which the user will not be able to modify, yet the bars will show the values been used for each frequency.

Table 21 provides a listing of all the requirements for the GUI.

Table 21 Software GUI algorithm requirements

Module	Algorithm	Requirement

GUI → State Detector	n/a	Receive input from Digital Equalizer
		Determine whether to activate EQ, update visualization, or update EQ screen.
		If current action is to activate EQ, notify Graphics Update so it interrupts any incoming visualization and stays ready for EQ graphics input.
GUI → Graphics Update	Visualization	Receive input from Graphics Generator I.
		Update LCD within 50 milliseconds.
		Listen for interrupt from EQ.
	EQ	Receive input from Graphics Generator II
		Update LCD within 50 milliseconds
		Ignore any interrupts from Idle.
	all	Maintain a flag indicating whether to ignore visualization.
		Send graphics to Display module.
		If EQ mode is active, immediately ignore any visualization graphics until Visualization mode is activated.
GUI → Visualization	n/a	Display visualization that represents the music playing.
		Must implement *Sleep*.
		Must implement *Wake*
		On wake, it executes infinite loop where it reads current characteristics of sound being played and utilizes them to generate visualization.
GUI → Visualization → Sound Analyzer	n/a	Must be able to read samples produced by EQ module.
		Must be able to generate an output that represents the input amplified based on

		equalizer values (stored in memory).
		Shall output the values needed for Graphics Generator I to produce graphics in a form of array stored in memory.
GUI → Visualization → Graphics Generator I	n/a	Must implement Sleep
		Must implement Wake
	Wake-up	On Wake Stay in an infinite loop until put to sleep.
	Awake	Shall read from memory the current characteristics of audio being played.
		Shall generate next graphic matrix.
		Notifies *Graphics Updater* when new graphic matrix is ready.
GUI → EQ	n/a	Must implement *Sleep*
		Must implement *Wake*
	Wake-up	Display EQ configuration screen.
	Awake	Update EQ configuration screen every time there is a user input.
		Receive input from EQ
GUI → EQ → Graphics Generator II	n/a	Listen to Values Updater.
		On wake, use current values to generate an updated graph for the Equalizer Control Screen.
		Notify EQ when graph is ready.
GUI → EQ → Values Updater	n/a	Update equalizer values based on touch screen values.
		Notify *Graphics Updater II* that there are new values.

4.12.2: System Design

The system will operate in two modes. One mode is the visualization created by an animated graphic which moves along with the amplitude, frequency and phase characteristics of the audio being played in order to provide a visual

representation of the music. The other mode is the equalizer mode which allows user to modify how the different frequencies are amplified. The programming modules will be programmed in C.

The animated graphics must have visualization which must represent the music post-amplification. The users must be able to identify in the visualization the different harmonics been played. Several visualization options may be added to the system later on should time and resources allow; however, only one is required. The visualization must update every forty to fifty (40 - 50) milliseconds in order to maintain real time graphics. This part of the system must receive input waves, analyze them, and generate output graphics fast enough for LCD controller to receive the next screen refresh and perform the update to the graphics within the stipulated time.

The equalization section of the system will provide the user a series of pre-determined equalizations as well as the ability to create a fully custom one. After fifteen (15) seconds of inactivity, it returns to visualization mode. The values which may be modified include:

Table 22 Software GUI System Design

Value	Description
Left/Right balance	User shall be able to modify the volume of the left and channels with respect to each other. This will be done through one visual control bar with a marker that slides left and right.
Six Frequency bars	This area will contain six bars where each represents a frequency.
	Frequency bars for Preset equalizations cannot be modified.
	Frequency bars for Customer equalizations can be customized by user via the GUI.
	Each bar can be set individually.
	Bars markers' move up and down only; where up it reaches the highest amplification value possible and down reaches the lowest.
Volume	Vertical bar which marker can be moved up or down.
	Highest volume is reached in the UP highest position.

Value	Description
	Lowest volume (mute) is reached in the DOWN lowest position.
Mute	On-screen button that sets volume to zero.
	If current state is mute, it will restore the volume to its value before it went mute.
EQ Presets	Allows user to select which equalizer configuration to use. There will be some pre-determined equalizer configurations, and user may choose “custom” in order to configure one of its own.
	Each pre-defined equalization as well as the custom ones, will be displayed as button on the screen.
Temperature	Display current temperature.
Source Select	There will be an on-screen button for each of the input sources.
	Only one can be active at a time.
Back	Switches from EQ mode to visualization mode.
Active Equalizer	Arrows that allow the user to switch between equalization presets, including two custom sets.
	The custom sets will provide the user the ability to adjust any of the frequency bars, and once a bar has been changed, the Equalizer Set settings will be automatically saved.
	The Pre-defined presets will not allow the user to modify any of the frequency values.
Shut-down	Initializes shut down sequence.
Display On/Off	Turns screen OFF

4.12.3: Test Plan

Each of the software components will be tested individually by the use of test algorithms that will simulate the input necessary for each module while the output is been captured by another test algorithm in order to ensure accuracy and correctness in functionality. After each software component has been tested individually, these will be combined with each other complying with the software diagram, and after every union, a test will be run. Where possible, the same test algorithms will be used; this will provide more security in the system’s correct

functionality. Once the all the components have been put together, they will be tested using a signal generator at a specific frequency to ensure all amplification channels are working, all visualization algorithms are producing the expected graphics, and that the EQ screen changes do in fact affect the output of the amplification and the visualization. Finally, each channel will be fed from a device of the kinds the final user is expected to use, as many different as possible will be used, in order to ensure system is performing effectively and efficiently in all aspects of the software.

Table 23 Software GUI Test Plan

Module	Test Plan
Graphics Updater	This module is the first one that will be coded; therefore, in order to test it, a “test function” must be used. Until other modules are ready to be incorporated, a series of images (backgrounds) will be passed on to this function in order to test whether it updates the LCD correctly.
Graphics Generator I	To test it, use this module to read a series of pre-defined values representing magnitudes of sound waves in decibels, and ensure graphics are generated correctly by using the same graphic for at least three seconds in order to notice whether the graphic generated truly matches the values, then reduce this until real time ensuring it maintains functionality and accuracy on the output graphs.
Graphics Generator II	The first test will be having this module display all the controls for the pre-defined Equalizer options, one where all bars are at 0, and one where they are all at maximum value.
Equalizer (EQ)	Pass on a series of values representing user input and ensure EQ updates values correctly and *Graphics Updater II* gets correct values.

Module	Test Plan
Touch Screen	Modify EQ values, wait for Idle to activate, and then activate EQ screen again and ensure values are kept. Also switch within the different pre-defined EQ sets and ensure they also stay configured.
Sound Analyzer	Pass a series of set values simulating Fourier Transform module output, and ensure it updates memory correctly. At this point, *Graphics Generator I* will be functioning; therefore, Sound Analyzer can be incorporated into the software in order to test it.
Values Updater	Turn all the bars down to minimum, and test one by one by raising them to their maximum.
	Using Oscilloscope, capture the output waves and ensure they are been amplified correctly while changing their values.
Idle	Using an oscilloscope, capture the output waves and compare them to the graphs being generated in the visualization.

4.13: Digital Potentiometer

Two SPI-bus-like bit-banger serial data streams are generated, with a common clock, reset, and enable lines. The two data streams are designated VR_SDI. Each VR_SDI signal is a 20-bit digital data stream and consists of two groups of 10 bits; the first group for U16 and the second group is for U15. Within each 10-bit group is a two-bit address and an eight-bit data value that indicates the "position" of the potentiometer being addressed. The VR_SDI signal is identical for both except for the volume control (VR8), which can be different in order to implement the left-right balance feature. Table 24 maps the digital potentiometer address to the function it controls.

Table 24

Channel	Function
VR1	30 Hz Equalizer
VR2	100 Hz Equalizer
VR3	300 Hz Equalizer
VR4	1000 Hz Equalizer
VR5	3000 Hz Equalizer

VR6	10000 Hz Equalizer
VR7	Pre Equalizer Gain
VR8 Left	Left Volume
VR8 Right	Right Volume

4.14: Analog to Digital Converter (ADC / Fourier Transform)

In our project we will be sampling a continuous signal with respect to time. We want to display the signal in the frequency domain were we would plot the input signal a graph of frequency vs. magnitude. We will be sampling the signal after the output of the equalizer in order to view the changes in the frequency response that the equalizer had done. This will give us a visual indication of how our equalizer is affecting the input signal and lets us tone accordingly.

Fourier transforms are a tool that enables a signal that is continues with respect to time to be transformed into the frequency domain. The Fourier transform can be simplified to the calculation of a discrete set of complex amplitudes, called Fourier series coefficients. Also, when a time-domain function is sampled to enable storage in a computer device, it is still possible to recreate a version of the original Fourier transform. Because computing the discrete-time Fourier transform from the mathematical definition is often too slow to be practical, so there was a method called the fast Fourier transforms that arose. A fast Fourier transform is a way to compute the same result more quickly. Computing the discrete-time Fourier transform of N points in the brute force way, using the definition, takes $O(N^2)$. While a fast Fourier transform can compute the same discrete-time Fourier transform in only $O(N \log N)$ operations. This time saving advantage of the fast Fourier transform is critical because we will be performing all the calculations real time on the Stellaris which also has to execute all the other operations that are happening.

There are numerous algorithms that enable the calculation of the Fast Fourier Transform. Our project implies some restrictions on which algorithm we can use because we have a limit amount of computing resources available. The Cooley Turkey is an example of an algorithm that we cannot use because of its method of operation. The is a heavy recursive algorithm and it relies of dynamic programming to reach the O(N log N) runtime. The problem with this is that due to our limited stack size we cannot support recursion because even at small values of N we will blow all the memory in our stack.

After researching for fast Fourier transforms algorithms that were designed to run on an embedded system a library that was written for ARM Cortex processors was found. Cortex M Software interface Standard (CMSIS) is a fully featured library that is by ARM for their processors. There is a specific version just for DSP that has over 60 functions including FFT which is what we will be using. The decision was made to use the CMSIS-DSP library over our own implementation

based on the fact that the CMSIS is highly optimized for the M4 that we will be using and will run far better than anything we can come close to making.

In the project design the requirements that were established was to have a 20 KHz max sampling frequency, and a minimal sampling frequency of 20 Hz. Another variable that must be considered is the FFT sampling size. The FFT size is a complex variable to pick. By picking a low FFT you increase your time response by decreasing the amount of samples that need to be taking which interns increases the refresh rate. But when the FFT size is small the frequency resolution is also lowered because the FFT size is the amount of frequency bins that you can divide the sampled data into. On the other side of the equation if the FFT size is increased to for example 4096 the time response decreases while the frequency resolution increases. Table 25 below gives all the relations that you will need to go back and forth from the time domain to the frequency domain.

Table 25

		Time Domain		
		F_s	N	T
Frequency Domain	$F_{Max}, \#\ lines, Df$	$\frac{F_{max}}{E_B}$	$\frac{\#\ lines}{E_B}$	$\frac{1}{Df} = \frac{\#\ lines}{F_{MAX}}$

The max sampling rate needs to be twice that of the input signal do to the Nyquist sampling theorem which implies the sampling rate to be at 40KHz. $F_s/N = b_{in}$ resolution, F_s is the input signal's sampling rate and N is the number of FFT points used If an FFT size of 2048 is decided there will be a frequency resolution of $19.54Hz/b_{in}$. The draw-back to this is that it will take .0512 seconds to collect the sample which implies a refresh rate of 19.53Hz.

One of the solutions to the slow refresh rate is a method that is implemented in software and it lets us feel the buffer twice as fast. This method is called multiple buffering, or ping pong buffering. In this method there is more than one buffer to take in the data from the reader. This lets the reader in our case the analog to digital peripheral to read in data at a higher rate. The optimal case would be for the ADC to read in data continuously. While one of the buffers is being filled the others can be processed. This eliminates the need to wait for the FFT to finish. Figure 29 is a flow block diagram that indicates the flow of the signal through the ADC and how the FFT transaction will happen.

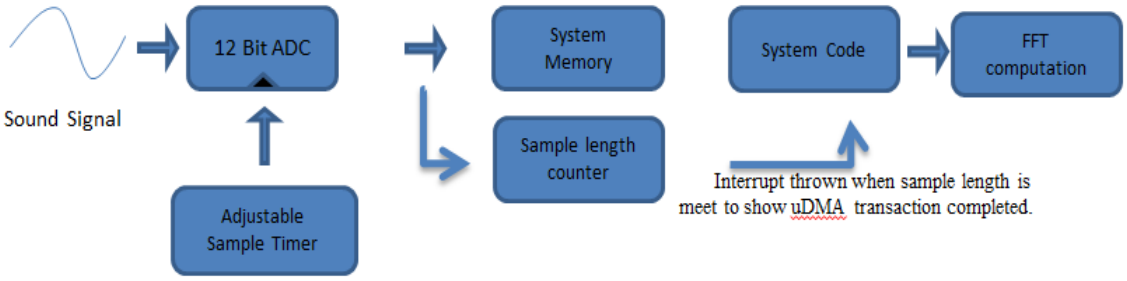


Figure 29

Table 26 below is a summary of the values that were calculated for use in the project. These values are subject to change once experimentation begins. Some of the values that are highly likely to change are the FFT size and the bin resolutions. This is because these values will directly affect the run time of the sampling algorithm and if it turn out that information is not being collected at a fast enough rate the first place to get a speed increase would be to decrease the FFT size which in turn will decrease the amount of iterations needed to collect each sample.

Table 26

Max input frequency	20KHz
Minimum input frequency	200Hz
Input bandwidth	19,800Hz
Max Sampling rate	40Khz
FFT sample Size	2048
Bin resolution	19.54Hz/bin
Sampling time	.0512s
Refresh rate	19.53Hz

4.14.1: Analog to Digital Converter

The MCU has two analog to digital converting units (ADC). Both ADCs share a total of 12 input channels. Each ADC is capable of obtaining a maximum sample resolution of 12 bits. Both of the ADC models are completely independent to enable them to throw independent interrupts and have different triggers. The way the sampling works on the microcontroller is that there are programmable sample sequences. Each sample has programmable variables such as input source; interrupt generating on completion, last sample indicator. Also each sequence can be programmed to start a µDMA transaction to efficiently move data from the FIFO of the ADC to the memory without the usage of the controller.

In the project the ADC will be triggered via an interrupt that is triggered by a timer that is set to a sampling rate of 40 KHz. The trigger source is defined in the

sample sequence. All triggers that are not being used are masked in by the ADC Interrupt Mask (AIM). After each sample sequence is complete the sampled data can be retrieved from the ADC Sample Sequence Result FIFO (ADCSSRFIFO) register. Once the data is here the ADC Sample Sequence FIFO Status (ADCSSFIFO) will show full. After each completed sample sequence a µDMA transaction will be triggered. The DMA controller will put the sample into the address which is implied in the DMA control register.

Because the ADC has voltage input range from the VDDA to GNDA and the analog signal source has a nominal signal range after the equalizer of ±1.5 volts peak, the audio signal must be conditioned to fall within the range of the ADC. U5A on the Analog Processor (see Figure 17) and the associated resistors provide the necessary level shift and bias.

5: Design Summary of Hardware and Software

This section discusses the processes by which all the hardware subsystems will be prototyped, laid out, fabricated and tested as subassemblies, then thoroughly tested.

5.1: Audio Processor

Most of Senior Design 2 was devoted to this design, of which there are two, A1 and A2. Many weeks were spent prototyping before the design was committed to copper then sent to the PC board manufacturer. This all happened before a mid-semester two week vacation by Stephen. Upon his return, the two boards were fabricated and tested. Figure 30 shows the final design of one of the assemblies.

Figure 30

5.2: High Voltage Power Supply (HVPS)

Because this assembly, A3, was needed to test the vacuum tube amplifiers, would be needed to test the Audio Processor, the prototyping of the Audio Processor had to be temporarily halted until the HVPS was built. Because the design is simple, it was directly built on a vector board. Figure 31 shows the final design. Unfortunately no suitable load resistor was located – it would have been 1.5 KΩ and at least 100 watts power rating. (None could be located available for free and its high cost precluded purchase as the project is way over budget.) Rather, it was tested with the Audio Processor.

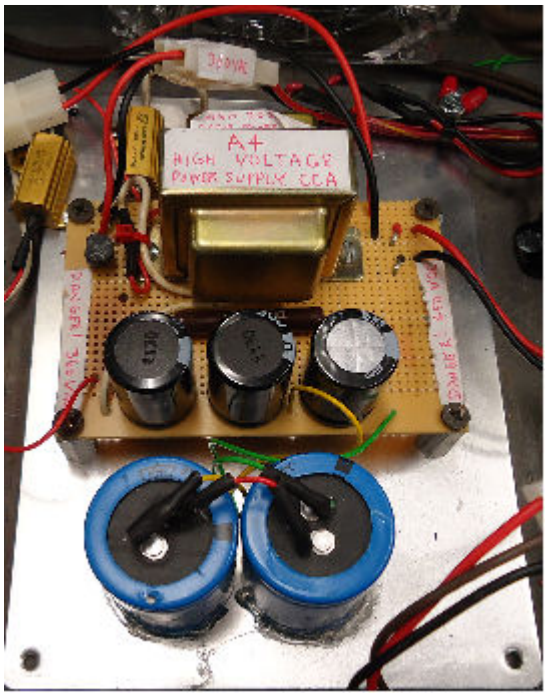


Figure 31

5.3: Audio Processor / HVPS Test

To be performed after the Audio Processor CCA (A1 and A2) is assembled, once for A1 and again for A2. Whichever is being tested will be referred to as the Unit Under Test (UUT) for the duration of this procedure. See Table 27 for the list of required test equipment. (Actual lab equipment used in parenthesis.)

PS1 DC power supply, capable of 6 volts at 2.5A (Agilent U8002A)
PS2 Dual DC power supply, capable of ±12 volts at 200mA (Leader LPS-152)
Audio signal generator with test leads in good condition (Tektronix AFG3022)
Digital multi-meter, with test leads in good condition (Tektronix DMM4050)
Digital Oscilloscope with probes in good condition (Tektronix MSO 4034B)
STM32F3 “Discovery” board, flashed with VR_Test test program
Banana-plug test wires as required
Female-to-female header jumper wires as required
4-way jumper (see Figure 33)
Voltage-Divider resistor (see Figure 33)
Dummy load: 8Ω / 20-watt resistor
Dummy load: 82 KΩ / 1-watt resistor
Fully tested High Voltage Power Supply A4 with T2
Modified AC power cord with mating connector for T2
Safety glasses for each test participant / observer
Insulated test surface
ESD-safe storage container for DIP integrated circuits.
500 to 1KΩ resistor

Solderless Breadboard
1KΩ to 10KΩ resistor

Table 27: Test Equipment List.

1. Inspect the UUT visually for workmanship. Measure all closely-spaced traces to verify that no zero-ohm continuity exists.
2. Set up the equipment as shown in Figure 33. Insure that the UUT is on an insulating surface and has no paper, plastic or any debris under it.
3. All test participants and observers must put on their safety glasses and leave them on during this procedure.
4. Measure continuity across the following connections on the UUT and verify that no zero-ohm continuity exists.
    a. J3 pins 1 to 2 and J3 pins 2 to 3
    b. J4 pins 1 to 2
    c. J5 pins 1 to 2
5. If any are installed, carefully remove the following parts from their sockets on the UUT with a gentle rocking motion and place them in the ESD safe storage container:
    a. U1 through U5, U11, U2, U15, U16, V1, V2 and V3
    b. The J6 jumper
6. Set PS1 to 0V with a current limit of 3.0A, voltage limit of 6.5 voltd, then disable the output.
7. Set PS2 for dual-tracking mode, +12V and -12V outputs ± 0.25V. If the supply is equipped with a current limit, adjust for 200mA maximum. Turn off the supply.
8. Connect PS2 as follows:
    a. +12V to J3 pin 1
    b. Ground to J3 pin 2
    c. -12V to J3 pin 3
9. Turn on PS1
10. Verify the following voltages (Measure using 500 to 1KΩ resistor, as the LM317 and LM337 regulators will not work correctly without a load (the ICs have been removed)
    a. +2.5 ±0.1 volts at J3-6 **Record here:_____**

    b. -2.5 ±0.1 volts at J3-7 **Record here:_____**

    c. +5V ±0.25 volts at J3-4 **Record here:_____**
11. Carefully install U11 and U12. Note that U11 is ESD-sensitive
12. Connect the oscilloscope to TP1 (U11-8)
13. Connect two terminals of the 4-way jumper to J1-8 and J3-5. This connects the analog and digital grounds together and stay on for the duration of the test. The remaining two terminals are used in the following test step.
14. For each input combination given in
15. Table 28:

a. Set the signal generator for the indicated sine wave signal outputs as close as possible (values given are as indicated by the AFG3022), then disable the output. For J3 pins 14 and 15, unplug the 4-way jumper for a “1”, and plug the 4-way jumper for a “0”.
b. Connect the signal generator “signal” wire as shown on the corresponding line and connect the “ground” wire to J1 pin 8 (ground).
c. Enable the output of the signal generator.
d. Measure the output on the oscilloscope and record in the appropriate place on Table 28.
e. The signal should measure approximately 2Vpp in each case.

Table 28

J3 pin 15	J3 pin 14	Signal Gen	Frequency	$V_{INPUT, PK-PK}$	$V_{OUTPUT, PK-PK}$
0	0	J1 pin 1	20 Hz	19mV (*)	
0	0	J1 pin 1	150 Hz	56mV (*)	
0	0	J1 pin 1	1 KHz	17mV	
0	0	J1 pin 1	4.5 KHz	39mV	
0	0	J1 pin 1	20 KHz	158mV	
0	1	J1 pin 3	1 KHz	1.0 V	
1	0	J1 pin 5	1 KHz	1.0 V	
1	1	J1 pin 7	1 KHz	100 mV	

* = Connect input signal using the dropping resistor assembly

16. Turn off PS2.
17. Connect scope probe to R7 end that is closest to U5 (U5-7).
18. Install U5 and jumper U16-18 to U16-19 with a 22-AWG wire inserted into the socket.
19. Set signal generator to 1 KHz and 100mV, and remove the 4-way jumper from J3 pins 14 and 15.
20. Turn on PS2, verify a 2Vpp signal on the oscilloscope then turn off PS2.
21. Install remaining semiconductors. Note that U15 and U16 are ESD-sensitive.
22. Connect Discovery Board to the UUT using the header wires per Table 29

Table 29

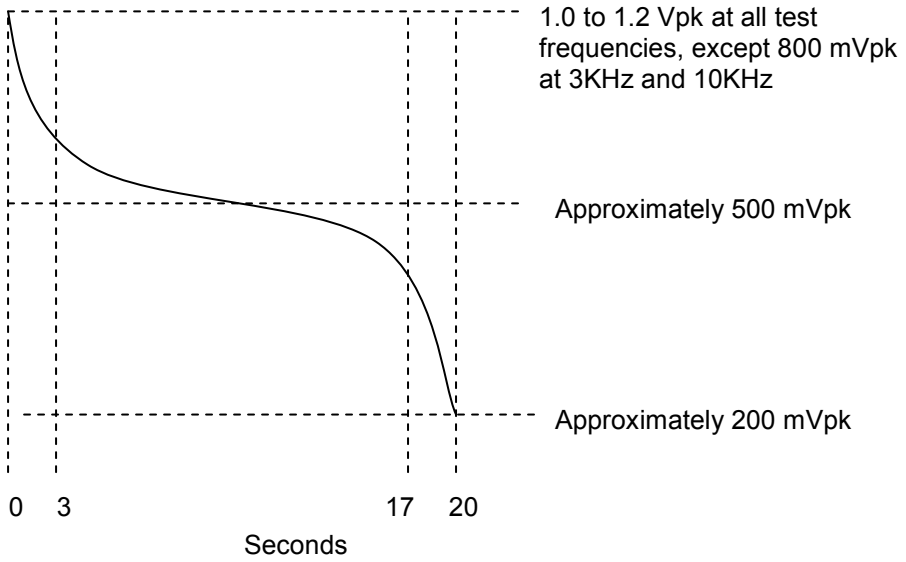
Discovery Board Connection	Discovery Board Label	UUT Connection	Signal
P2-1	+5V	J3-6	+2.5 volts
P1-43	PD13	J3-12	VR_SDI
P1-44	PD12	J3-8	VR_nCS
P1-45	PD15	J3-9	VR_nRS
P1-46	PD14	J3-13	VR_CLK
P1-50	GND	J3-7	-2.5 volts

23. Turn on PS2. It may be necessary to boost the current limit, but in no case above 200 mA draw. Be aware that U18 and U19 may get hot, so temporarily attach a *separate* alligator clip to each. Do not connect them together or their outputs will become shorted.
24. Verify that LD2 on the Discovery board flashes at about a 1 Hz rate. This verifies correct operation, and must be investigated before continuing to avoid damage to the Discovery or the UUT.
25. Test the VR chip functionality as follows, using the data in Table 30 and Figure 32:
    a. Connect the scope to the test point indicated.
    b. Set function generator as indicated.
    c. Push the blue button on the Discovery board until the indicated LED is lit.
    d. Verify the output signal behaves as indicated. Note that this behavior has a 20-second period and you much watch the signal carefully throughout. Ignore any glitches (they are caused by rapidly-changing Discovery board signals), but investigate any deviation before proceeding

Table 30

Function	**Scope**	**Generator**	**LED**	**Output**
Pre-amp gain	R7, closest to U5	1 KHz ,100 mVpp	LD8	Linearly increases from 0 to 2Vpp
30 Hz EQ	TP2	30 Hz, 60 mVpp	LD4	Amplitude profile as shown in Figure 32
100 Hz EQ	TP2	100 Hz, 60 mVpp	LD3	
300 Hz EQ	TP2	300 Hz, 60 mVpp	LD5	
1 KHz EQ	TP2	1 KHz, 60 mVpp	LD7	
3 KHz EQ	TP2	3 KHz, 60 mVpp	LD9	
10 KHz EQ	TP2	10 KHz, 60 mVpp	LD10	
Volume	TP2	1 KHz ,100 mVpp	LD6	Linearly increases from 0 to 2Vpp

Figure 32



26. Turn off PS2.
27. Remove the Discovery board and the jumper connections to the UUT.
28. Jumper a pull-up resistor of between 1 and 10 KΩ between J3-6 and J3-9.

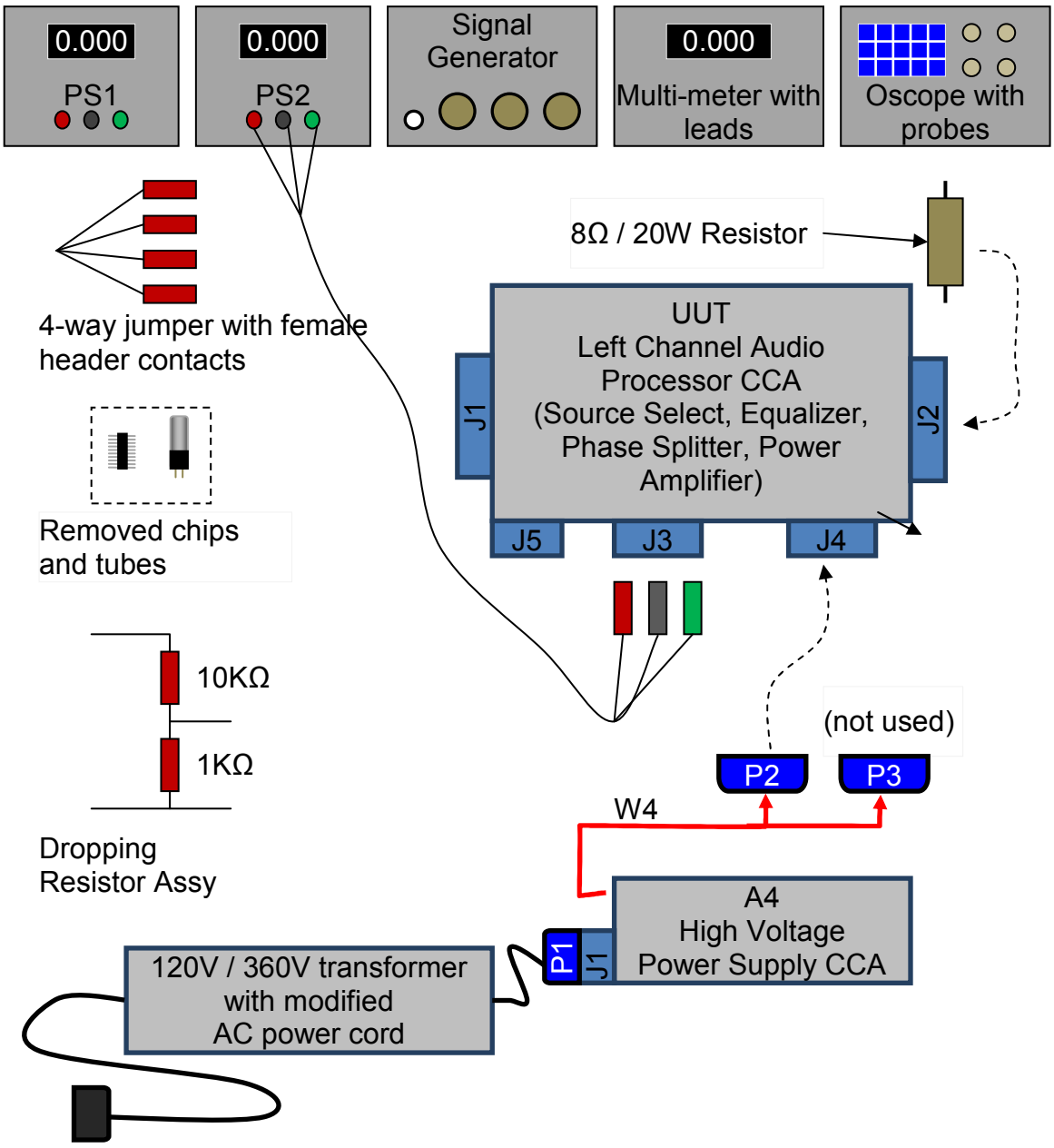
EXTREME DANGER - - - SHOCK HAZARD!
The A4 power supply generates 450 volts at 300mA, and is capable of storing a potentially lethal charge for several minutes even after being removed from the 110 Volts AC power source. Use only one hand to make measurements – keep your other hand away to avoid a shock through your chest. Follow all test steps exactly – do not deviate. If any unexpected results are encountered, unplug the power supply from the AC power source and wait a minimum of three minutes before any attempt to inspect the CCA is made. Another person should be on hand to render aid if necessary.
SHOCK HAZARD EXISTS FOR THE REST OF THE THIS PROCEDURE
CONTINUE TO WEAR SAFETY GLASSES
KNOW THE LOCATION OF A FIRE EXTINGUISHER

29. Connect the 82KΩ / 1W dummy load from TP-8 (D1-Cathode) and GND test point on the UUT.
30. Set the multi-meter to the **1000 VDC range**, and connect the black lead to the ground bus on the CCA and the red lead to R57 top end. Insure that both leads are secure, and can't touch anything else by insulating them with electrical tape. **DO NOT TOUCH while power is applied.**
31. Plug the Modified AC power cord into a source of 110 VAC. **The measurement in the next step should be made within one minute to prevent the nearly unloaded HVPS from reaching 500 volts.**
32. This voltage is the output of the High Voltage Power Supply. It is unregulated and can be as high as 500 volts when the tubes are cold, but will lower as the tubes heat up, ultimately settling around 410 volts. **Record here:_____**

33. Unplug the modified AC cord and **wait until the multi-meter reads zero** before continuing
34. Move the red multimeter lead to D1-Cathode. Insure that both leads are secure, and can't touch anything else. **DO NOT TOUCH while power is applied.**
35. Plug the Modified AC power cord into a source of 110 VAC. **The measurement in the next step should be made within one minute to prevent the nearly unloaded HVPS from reaching 500 volts.**
36. This voltage is plate supply voltage for V3. It is not fully regulated, but rather is prevented by D1 through D3 from becoming excessive. Verify a voltage of 225 to 250 VDC. **Record here:_____**
37. Unplug the modified AC cord and **wait until the multi-meter reads zero** before continuing.
38. Carefully reinstall the vacuum tubes. Do not force in to their sockets, rather use a circular rocking motion on each tube until it is fully seated.
39. Connect the 8-ohm resistor to J2-1 and 2. Connect the oscilloscope input across the terminals of the resistor
40. Set the signal generator to 1.25KHz but turn the output level control to 10 millivolts.
41. Set the multi-meter to the **1000 VDC range**, and connect the black lead to the ground bus on the CCA and the red lead to R57 top end. Insure that both leads are secure, and can't touch anything else by insulating them with electrical tape. **DO NOT TOUCH while power is applied.** Continue to monitor this voltage, which should settle in the 400 to 420 volt range once the tubes warm up.
42. Connect W4P2 to UUT J4. Fully insulate the unused W4P3 with electrical tape.
43. Verify all connections!
44. Turn on PS1. Enable the output, then slowly ramp-up the voltage over a period of at least 60 seconds to 5.5 volts. Verify a current draw of approximately 2.4 amps. (PS1 powers the tube filaments, which start with a very low resistance that increases as they heat. This method must be followed each time the tubes filaments are to be powered by PS1.)
45. Turn on PS2.
46. Plug the Modified AC power cord into a source of 110 VAC. **Be ready to unplug in an emergency if anything smokes or explodes.**
47. Watch the tubes as they "warm up". They may light with a dim orange glow, and must never get obviously bright. Allow one minute for full warm up.
48. Slowly increase the amplitude of the signal generator while watching the oscilloscope. 300mVpp is the maximum input before the VR chips clip due to the 10X gain, beyond that the signal distorts strongly.
49. A very clean sine wave should be observed of about 11Vpp. Check distortion by using the scope's FFT function and enter the amplitudes of the first five harmonics into a suitable spreadsheet.

50. Calculate the Root Mean Square voltage across the 8Ω load and **Record here:_____**
51. Calculate the power across the 8Ω load and **Record here:_____**
52. If time is available, allow the equipment to remain in operation for **1 HOUR**, to act as a burn-in test. Monitor the voltage and current readings, and note how hot the power supply is getting. Unplug the modified AC cord if any reading suddenly changes or the power supply begins to smoke or explode.
53. Unplug the modified AC cord and **wait until the multi-meter reads zero** before continuing.
54. Turn off PS1 and PS2
55. Reinstall U4
56. Disconnect all equipment – TEST IS COMPLETE.

Figure 33: Test Setup. (Created by the authors)



5.4: Audio Processor Testing

After each Audio Processor CCA was built and tested as described in paragraphs 5.1 and 5.2, testing was conducted to determine its maximum power output capability and it's THD at various power levels. The Tektronics AFG3022 generator and Tektronix MSO4034B oscilloscope with appropriate test leads and probes were used. Unless noted, all testing was performed using a 1.25KHz sine function, as this frequency produced harmonics that lined up with the graticules of the oscilloscope. For consistency and ease of test, only the fundamental and harmonics up to fifth order were measured. The amplitude of higher order

harmonics are assumed to be very small, however if they were included, the true THD would be somewhat higher than indicated in each test. In all FFT photos to be shown, the 1.25 KHz fundamental is at the left, the 5th harmonic is in the center and the 9th harmonic is at the right.

5.4.1: <u>Baseline</u> Figure 34 is an FFT photo of the output of the generator alone, when set to "300mVpk". Only odd harmonics are generated, consistent with the all-solid-state design of the generator, yielding about 0.6% THD.

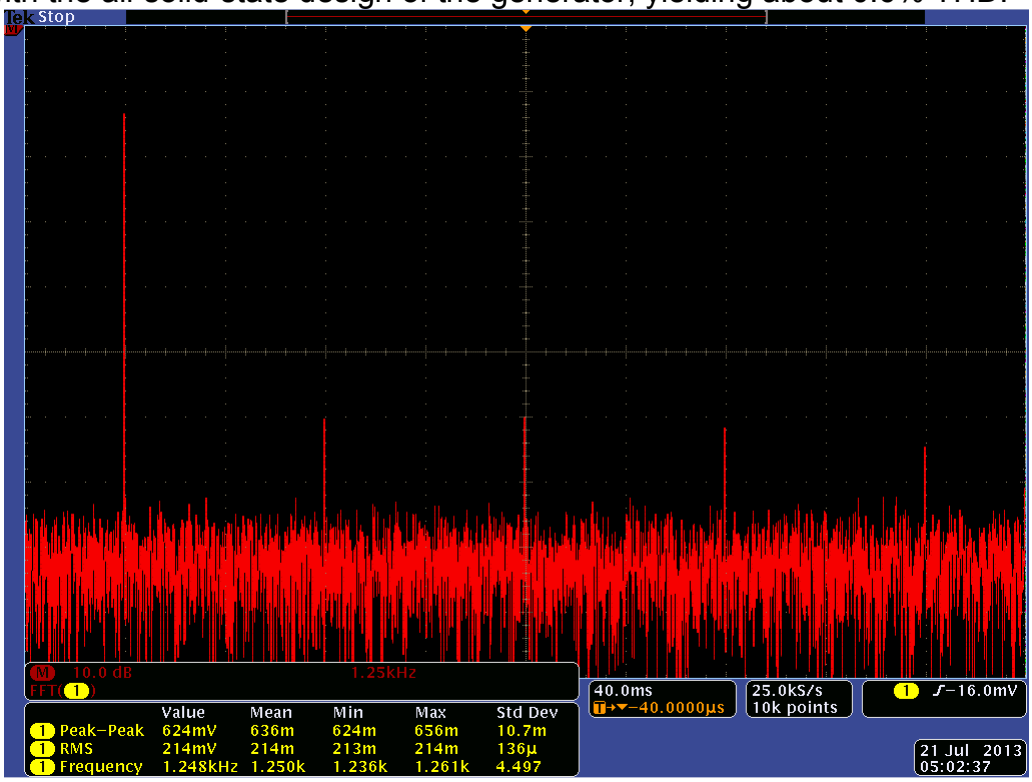


Figure 34

5.4.2: <u>A1 distortion tests</u>

Table 31 provides the result of distortion measurements performed on the A1 Audio Processor. Power output and THD were measured across a 7.5Ω test resistor that was available. For this test, the solid state circuits were bypassed and the generator output was applied to TP2. The amplitude of the generator was adjusted to yield the indicated power. The THD is higher than desired, however is consistent with the prediction that THD increases with power.

Power Output (rms)	THD measured
100 mW	2.1%
200 mW	1.2%
500 mW	2.0%
1 W	1.8%
2 W	2.0%
5 W	2.3%
10W	3.4%

Power Output (rms)	THD measured
20 W	6.0%

Table 31

Figure 35 shows the FFT photo taken during the 1 Watt test. Compared with Figure 34, note the addition of the 2nd order harmonic, consistent with that expected of a vacuum tube amplifier. Note that the 3rd and 5th order harmonics are not appreciably higher than the generator alone, implying that the true THD of the vacuum tube amplifier is lower than the results that Table 31 would seem to indicate.

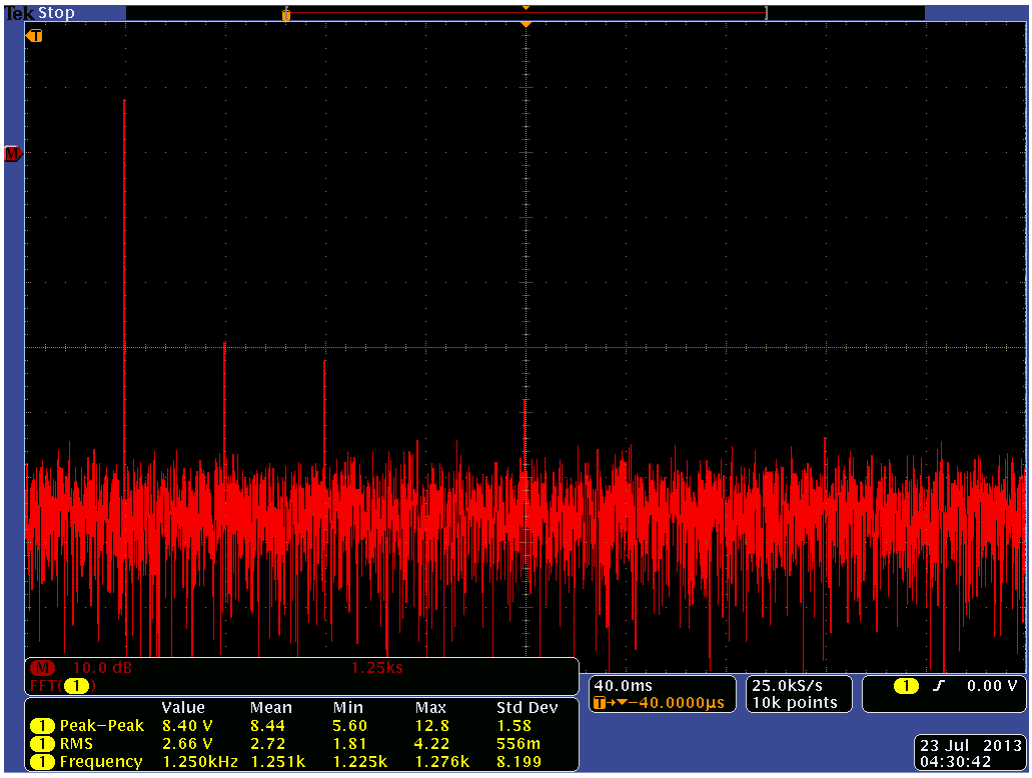


Figure 35

Table 32 is a spreadsheet calculation made during the 1 watt test, and is representative of that used to test all power levels.

	THD Calculator		
Date of test	7/22/2013		
Heaters	5.5VDC	2.26A	From power suppl
B+	419	VDC	As indicated on mu
Load	7.5	ohms	resistor value
Input	50	mVpp	As indicated on ge
Output	2.67	Vrms	As indicated on sc
Pout	1.0	Watts rms	
	OUTPUT OF AMPLIFIER		
Harmonic	**Amplitude (dB)**	**Delta from 1st (dB)**	**Delta from 1st (V)**
1	8.00	0.00	1
2	-29.00	-37.00	0.014125375
3	-32.00	-40.00	0.01
4	-60.00	-68.00	0.000398107
5	-38.00	-46.00	0.005011872
	THD	1.8%	

Table 32

5.4.3: A2 maximum power test

For this test, the signal generator was connected to the “AUX” input on J1 of the Audio Processor, and adjusted to yield the highest output without visible clipping as measured across a 7.5Ω test resistor. The result, shown in Figure 36, is 13.3 Vrms corresponding to an astounding 23.4 watts rms across the resistor. A measurement of THD was performed; an FFT photo of the result is shown in Figure 37. Note the moderate levels of even-order harmonics. The harmonic calculation (not shown) yields an astonishing low 1.6% THD

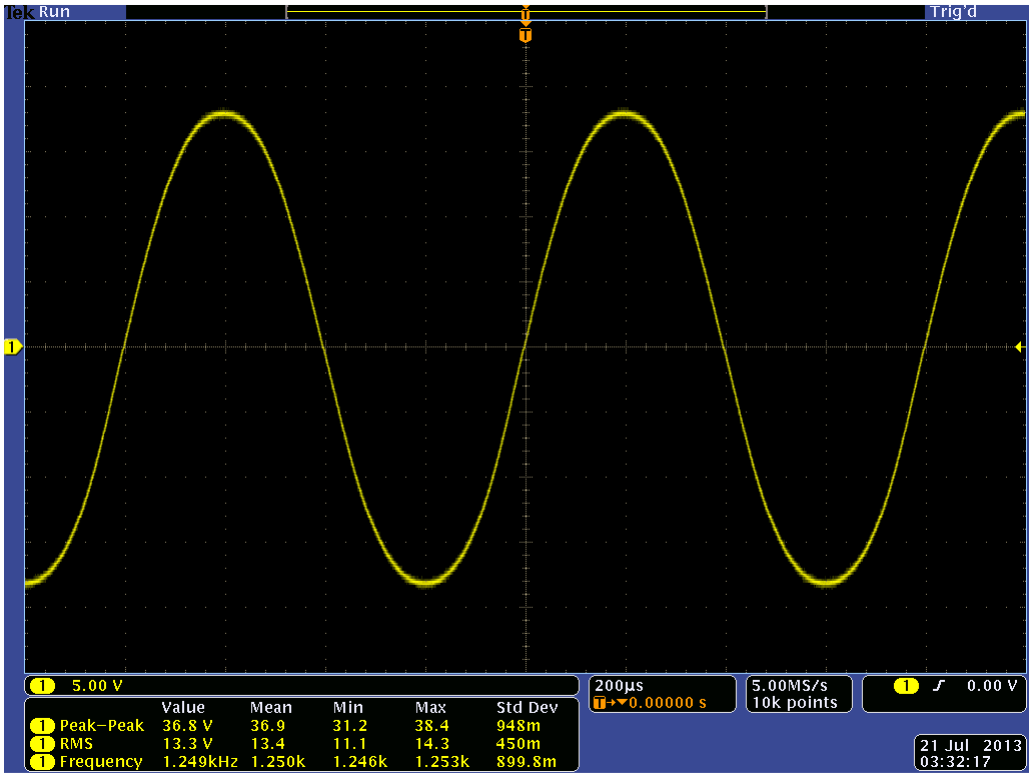


Figure 36

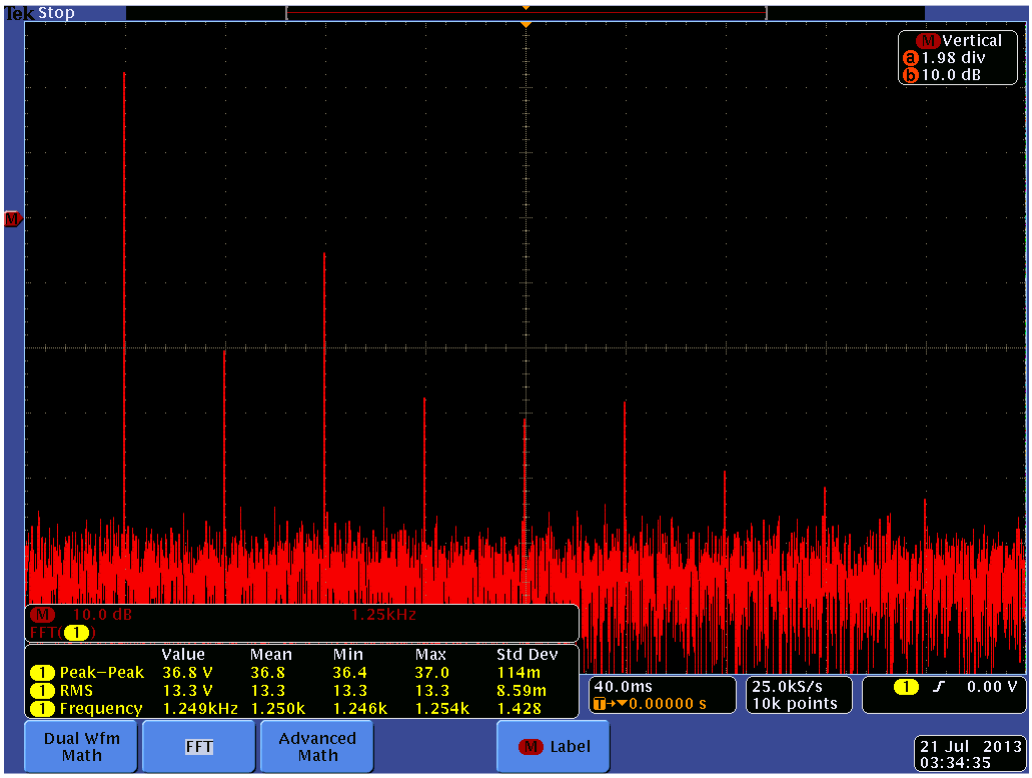


Figure 37

5.4.4: A2 frequency response test

This test was made last, and unfortunately after the A1 card suffered an overload that caused the tubes and output transformer to get extremely hot. Afterwards the A1 card was never able to reach its maximum output level. Since this occurred in the day before the presentation, with replacement parts unavailable, an investigation was deferred until after Senior Design 2. This effect was noticed by the audience as a reduced amplitude from the left speaker.

The frequency response of both A1 and A2 was measured, but the A1 card now suffers from anomalously low low-frequency response. Accordingly, only the results of the A2 card test will be presented. For this test, the signal generator was set to produce a sweeping tone from 20 Hz to 20 KHz over a 600 mS period, (which allows easier estimation of frequencies from the oscilloscope display) and connected to the AUX on J1. A separate “sync” output from the generator was connected to the oscilloscope to create a stable display, and finally the output taken across the 7.5Ω test resistor was connected to the oscilloscope for display.

The result, shown in Figure 38 shows input signal on channel 1 (yellow), the sync signal on channel 2 (cyan) and the output on channel 3 (magenta). The frequency response can be determined by inspection of the peak amplitude of channel 3, 20Hz on the right, 1 KHz at 300mS, and 20 KHz at 600 mS (the source of the gap at about 550 mS is unknown). The frequency response shows a mild roll-off at the low end and a mild roll-up at the high end. Using 2V at 1 KHz as the reference amplitude, the 1.6V at 20 Hz is 1.9dB down, and the 2.4V at 20KHz is 1.9dB down. A second test was made, this time taking the output from TP2 (just prior to the tube amplifier), shown in Figure 39, shows a nearly flat response, meaning that the frequency response errors originate in the vacuum tube amplifier. This result, while within specifications, is unexpected and due to the timing will need to be investigated after Senior Design 2.

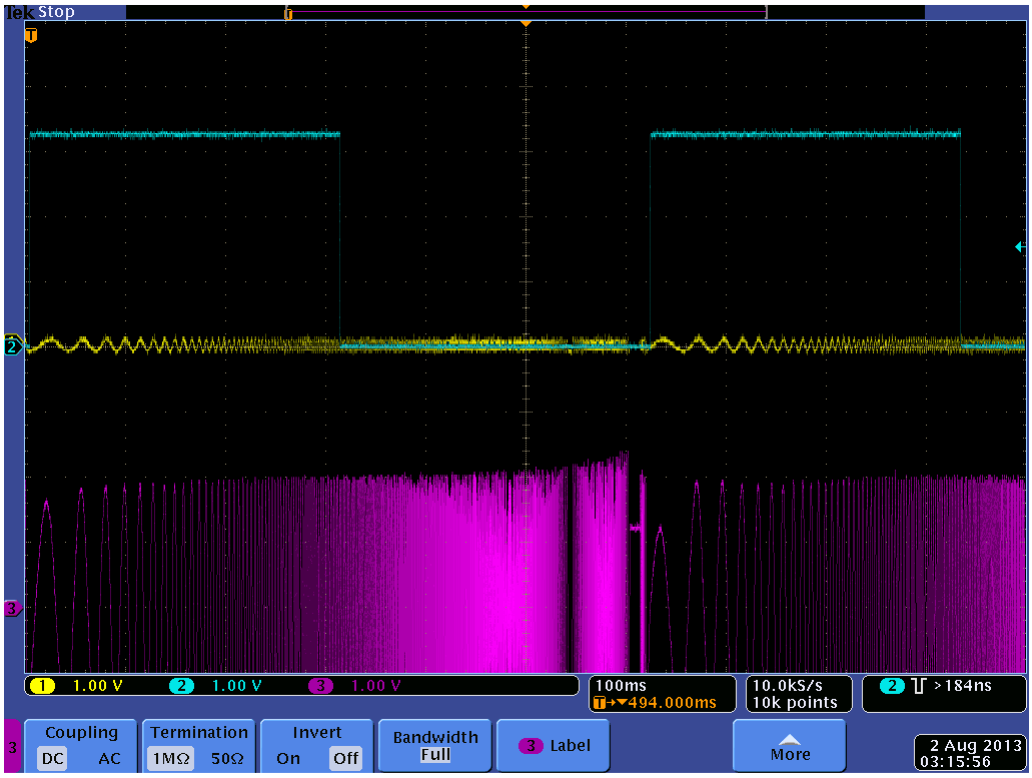


Figure 38

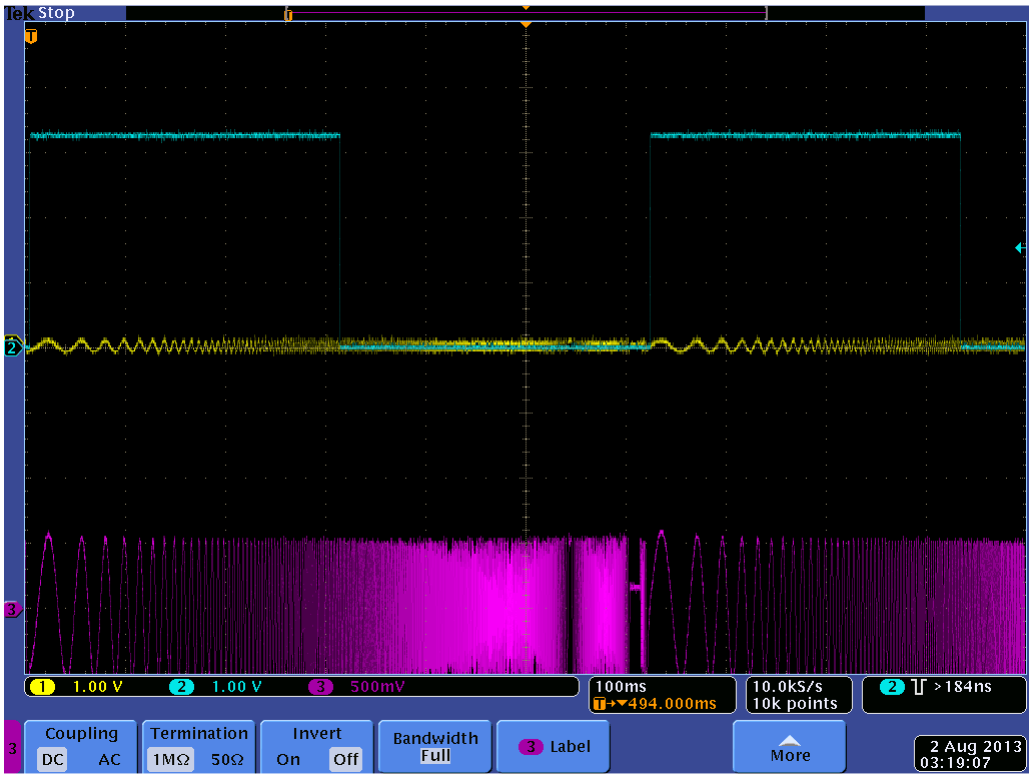


Figure 39

5.5: Microcontroller

This assembly, A4, was not prototyped. Rather, a design was created based upon a reference design provided by STMicroelectronics – see paragraph 4.8. The STMicroelectronics STM32F303VCT6 is a 100-pin LQFP, the only surface-mount part in the project. Someone from UCF's Amateur Radio Club was enlisted to solder this part, after which the remaining parts were soldered by the authors. Figure 40 shows the as-designed assembly. Figure 11 shows the as-built assembly in the lower-center, but the assembly is too buried to show well.

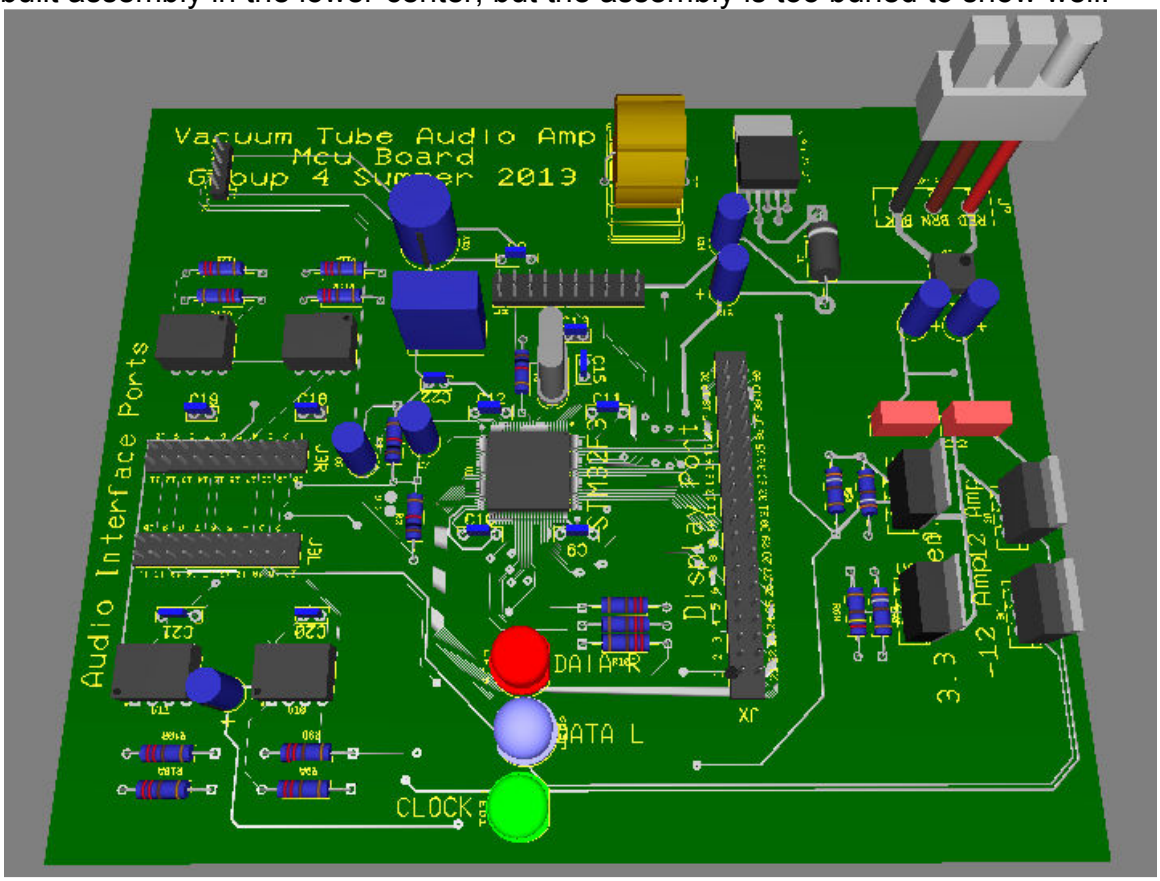


Figure 40

5.6: Microcontroller Testing

Testing was performed using the application software, which was able to implement some of the command mode functionality of Figure 5: The currently-selected button on the LCD was able to be moved and the volume could be changed. Unfortunately there was not enough time to implement all features. When no touch occurred for more than several seconds, a visualization of jumping bars whose maximum height corresponded to the maximum audio signal level was observed. Finally, any touch returned to the command mode.

5.7: Final hardware checkout.

The following tests were intended for final verification of the finished product, but never ran due to time constraints.

TEST: Verify that all exposed metal parts have continuity to the ground terminal of the power cord. The maximum resistance to any point shall be less than 1.0 ohms.

TEST: Bandwidth: 20 Hz to 20 KHz flat ±3dB. This measurement will be performed using a 1000 Hz sinusoidal signal set to a specific, TBD voltage level as an input to the amplifier. The volume will be set to a TBD voltage level and a reference measurement made. The measurement will be repeated at a large sample of frequencies covering the audio spectrum. The amplifier output terminals will be connected to an 8-ohm resistive load.

TEST: Total Harmonic Distortion: 0.5% maximum, measured at a number of frequencies 100 Hz to 5 KHz, 12dB below maximum output.

TEST: Thermal rise from inside the cabinet

5.8: Deleted Features

During the final design and fabrication process performed during the Senior Design 2 semester, some of the features were deemed to be no longer required and accordingly were removed from the design. Table 33 lists the deleted features.

Table 33

Feature	Paragraph discussed in the Senior Design 1 document	Reason for removal
Speaker Relay	1	Turn on transients were found to be a lot smaller than anticipated; below the level that would be destructive to speakers
Software controlled power sequencing	3.4.3, 4.11	See paragraph 5.10.1

5.8.1: Software controlled power sequencing

During the initial research performed, the documentation for the LCD was very poor and contained conflicting information. Among other things, it appeared that up to six different voltages were required that had to be turned on and off in a

specified sequence. Once the LCD arrived, it was subsequently determined that only two 3.3 volt sources were required which were not time critical. Since there was no longer any need for the sequencing controller, it was decided to eliminate the main power switch bypass relay to simplify the design.

5.9: Deferred Features

Due to a lack of time in the Senior Design 2 semester, which is only ten weeks long, many of the desired features were not implemented as described in this document. Table 34 lists the deferred features. These features will be implemented after Senior Design 2 on a hobbyist basis.

Table 34

Feature	Paragraph discussed in the Senior Design 2 document	Reason for deferral
Touch screen	2	The touch screen that was purchased was found, after extensive effort was expended, to not work reliably. In the last week of Senior Design 2, when it was too late to order a replacement, the decision was made to not use the touchscreen for control. A digital touch interface panel was quickly constructed and integrated as an alternative control feature.
"finished cabinet of contemporary design"	2.3.1 and 3.8	Too difficult to implement during the limited time available during Senior Design 2
Top-mounted tubes	2.3.1	Too difficult to implement during the limited time available during Senior Design 2
Shielded transformers	2.3.1	Lack of time to research sources of Permalloy 80
Shielded sub enclosure	2.3.1	Lack of time to research sources of Permalloy 80
Internal chassis temperature measurement	2.3.1	Overlooked until it was too late to add. No research of suitable temperature sensors was ever performed
Toggle-type power switch	2.3.1	No source of free stock was available. Part that was finally ordered arrived too late to be used for the Final Presentation.
IEC-320 C13 power connector	2.3.1	No source of free stock was available

Feature	Paragraph discussed in the Senior Design 2 document	Reason for deferral
The use of twisted shielded pair wire	2.3.1	Cost savings – no source of free stock was available
Digital circuits in shielded sub enclosure	2.3.1	Too difficult to implement during the limited time available during Senior Design 2
Splash screen	2.3.2	Too difficult to implement during the limited time available during Senior Design 2
Three to six visualizations	2.3.2	Too difficult to implement during the limited time available during Senior Design 2
Insulating barrier over the terminal strip	4.10	Overlooked
Full-feature implementation in the software	5.4	Too difficult to implement during the limited time available during Senior Design 2

5.10 Design changes

Table 35 lists the design changes were made to the design during Senior Design 2.

Table 35

Location	Change	Reason
Input Source Select	Add phonograph ground screw	For phonographs that provide a ground screw, this connection greatly reduces 60 Hz hum and buzz sounds
Graphic Equalizer – analog portion	Volume potentiometer moved to be before the final op-amp buffer that generates the AUDIO_EQ signal	Vacuum tube phase-splitter is less sensitive than predicted, and requires nearly 8Vpp for full output. Since signals passing through the digital potentiometers are limited to 5Vpp, this change allows the signal to be boosted sufficiently.
Graphic Equalizer – analog portion	Resistors added to form the Level Shifter and Bias stage, resulting in the AUDIO_SAMPLE signal	Design completion (values were not known at the completion of Senior Design 1)
Project-wide	Some components were changed in value	Due to availability, space constraints on the PCB or performance reasons

5.11: Specification Compliance

Table 36 lists all the requirements along with the actual value achieved. In general the results indicate fair-to-good compliance.

Requirement	Value Desired	Value Achieved
Number of audio channels	2 (stereo)	2 (stereo)
Output power rating	10 Watts Root Mean Square (RMS)	23.4 Watts Root Mean Square (RMS)
Input impedance	10 KΩ	47 KΩ for the phono input; 10 KΩ for all others
Output impedance	8 Ω	8 Ω, however was demonstrated driving a slightly lower impedance during tests
Bandwidth	20 Hz to 20 KHz flat ±3dB	20 Hz to 20 KHz flat ±1.6dB
Total Harmonic Distortion, low signal level	0.5%	1.2 to 2.0%
Total Harmonic Distortion, high signal level	2.5%	3.4% at 10 watts rms “rated” power; 6.0% at 23.4% maximum power
Hum	None detectable	Some hum, especially when the phono input is used.
No ungrounded exposed metal surfaces	Full compliance	

Table 36

5.12: Difficulties

Table 37 lists the difficulties that arose during testing and the mitigation required to overcome them.

Issue	Mitigation
Any attempt to display pure white on the LCD causes excessive current flow, and often causes the regulators on the LCD panel to shut down	Do not display white, especially over large areas of the display
It takes longer than anticipated to refresh the entire LCD	Refreshing will be performed over the smallest possible area

Issue	Mitigation
Touch screen is unreliable, generates incorrect results or fails to respond to touch at all.	Touch screen is deemed to be defective. As a temporary measure, a digital touch interface (the small red panel below the LCD) was added for demonstration. Ultimately, the touch screen must be replaced.
Voltage regulators get too hot, particularly U5.	Heat sinks will be added after Senior Design 2
AD8403 chips are extremely ESD and improper-voltage sensitive, resulting in numerous mishaps	Follow basic ESD protocols when inserting or removing these chips and double check all rectangular connectors before applying power
Distortion is a little higher than predicted	The signal generator was found to produce its own distortion, up to 1% THD with a "pure" sine function. The harmonics are all odd-order and should be subtracted from the THD of the Audio Processor during test. That is, use FFT(Audio Processor)-FFT(Generator) to calculate distortion.
Frequency response of the tube amplifier is within specifications, but not flat as indicated during simulation (see 5.3.4)	Will be investigated after Senior Design 2.
Hum is excessive	An extra 500µF of capacitance was added at the output of the HVPS. The DC ground was connected to the AC earth ground terminal. A ground wire must be connected from the phonograph to the DC ground. These measures help, but do eliminate the hum, and will be investigated after Senior Design 2.

Table 37

5.13: Future Plans

From the beginning it was the intention of team member Stephen to take ownership of the finished project upon completion, and use it for personal entertainment. But development of this project will continue as a hobby for Stephen in coming years. The paragraphs below discuss some of the upgrades and design changes that are already anticipated

5.13.1: Input wiring

For presentation purposes, and because the U11 chip on both A1 and A2 was damaged without a replacement available, only the phonograph was wired for test. All defective ICs will be replaced and all inputs will be wired. Additionally, a power switch will be added.

5.13.2: Damage mitigation

J3 on both Audio processors and J3L and J3R on the MCU CCA will be replaced with keyed connectors to prevent the mismating of the ribbon cables that caused most of the damaged parts during development.

5.13.3: Improved graphics

The version of the software used to demonstrate the project only provides a single visualization with no background images, so additional visualizations will be developed and will make use of the SD card reader to store bit-mapped graphics that can be used to enhance the display.

5.13.4: Hardware diagnostic

A testing mode will be implemented that can be entered by jumpering a TBD GPIO pin to ground, and be used to troubleshoot suspected malfunction. Various submodes will be created: 1) The GPIO outputs will generate known waveforms that can be tested using an oscilloscope. 2) Fill the LCD screen with color bars. 3) Generate an ASCII terminal readable version of the touchscreen/

5.13.5: Hum mitigation

Permalloy shielding will be added around all transformers, and the MCU. AC wiring will be replaced with twisted shielded pair wires.

5.13.6: DC power distribution

The DC voltage regulators on A1, A2 and A4 will be removed. A new low-voltage power supply assembly will be designed and fabricated, with all voltage regulators adequately sized and heat sinked.

5.13.7: New Cabinet

A generation 2 cabinet will ultimately be fabricated; the anticipated design is shown in Figure 41.

Figure 41: Generation 2 cabinet. (Created by Joshua Nichols, son of one of the team members)

6: Administrative Content

Project member Stephen was the team member who took care of administrative content.

6.1: Project Milestones

Table 38 lists the milestones of this project.

Week	Event
SD1, week 3	Project selected from all ideas presented between the team members
SD1, week 7	Power amplifier trade study complete, and design topology chosen
SD1, Week 11	Parts procurement began, graphic equalizer design complete
SD1, Week 14	The LCD/Touchscreen arrived. This part was ordered from Hong Kong and took 30 days to arrive; the longest lead time in the project
After SD1, before SD2	Finalize design, order parts
SD2, Week 1	(Completion of SD1) Begin prototyping of the Audio Processor. Software development accelerates
SD2, Week 2	Begin design and build of the HVPS
SD2, Week 4	Critical Design Review
SD2, Week 5	Layout and order Audio Processor PCB
SD2, Week 6	Layout and order MCU
SD2, Week 7	Progress Demo to Dr. Ritchie
SD2, Week 8	Fabrication of all CCAs begins
SD2, Week 9	First CCA is tested; others are tested as finished
SD2, Week 10	Final Assembly and Test

Table 38

6.2: Budget vs. Actual Cost

The budget for this project is $500. The actual cost was not fully tracked but is believed to be approximately $800. Approximately $50 of parts were ordered and not used, and approximately $50 were damaged (mostly U11, 15 and 16 on the Audio Processor) during test.

6.3: Bill of Materials

Table 39 comprises the Bill of Materials for the electrical portion of this project, as of the completion of Senior Design 1. Note the following subassembly codes:

AP/A = Audio Processor Part A
AP/B = Audio Processor Part B
HVPS = High-voltage power supply
ISS = Input Source Select
VTA = Vacuum Tube Preamp / Amp
PS/M = Power Supply / Microcontroller

All resistors 1/4 watt unless noted

Assy	Sub Assy	Ref Des	Qty per Assy	Total Qty	Manu-facturer	Part Number	Description
A1/A2	ISS	J1	1	2			4X2 Header Rcpt, .1" spacing, board mount
A1/A2	ISS	C11	1	2			100pF ceramic cap
A1/A2	ISS	C12	1	2			100uF / 16V electrolytic cap
A1/A2	ISS	C13	1	2			4.7nF ceramic cap
A1/A2	ISS	C14	1	2			15nF ceramic cap
A1/A2	ISS	C16	1	2			33uF / 16V electrolytic cap
A1/A2	ISS	C17-C19	3	6			1uF / 25V tantalum cap
A1/A2	ISS	R11	1	2			47K resistor
A1/A2	ISS	R12	1	2			390Ω resistor
A1/A2	ISS	R13	1	2			16KΩ resistor
A1/A2	ISS	R14	1	2			200KΩ resistor
A1/A2	ISS	R16, R18-R21	5	10			10KΩ resistor
A1/A2	ISS	R17	1	2			100KΩ resistor
A1/A2	ISS	U11	1	2	Analog Devices	ADG408BN	8-1 Analog mux, DIP
A1/A2	ISS	U12	1	2	TI	LM4562NA	Hi performance dual op amp, DIP
A1/A2	AP/A	C1a	1	2			.1uF /16V tantalum cap
A1/A2	AP/A	C1b	1	2			10uF /16V tantalum cap
A1/A2	AP/A	C2a	1	2			0.039uF /16V tantalum cap
A1/A2	AP/A	C2b	1	2			2.2uF /16V tantalum cap
A1/A2	AP/A	C3a	1	2			0.01uF /16V tantalum cap
A1/A2	AP/A	C3b	1	2			1uF /16V tantalum cap
A1/A2	AP/A	C4a	1	2			3900pF / 16V tantalum cap
A1/A2	AP/A	C4b	1	2			.22uF /16V tantalum cap
A1/A2	AP/A	C5a	1	2			1000pF / 16V tantalum cap
A1/A2	AP/A	C5b	1	2			0.1uF /16V tantalum cap
A1/A2	AP/A	C6a	1	2			390pF / 16V tantalum cap
A1/A2	AP/A	C6b	1	2			0.022uF /16V tantalum cap
A1/A2	AP/A	R1a, R2a, R3a, R4a, R5a, R6a	6	12			62KΩ resistor
A1/A2	AP/A	R1b, R2b, R3b, R4b, R5b, R6b	6	12			470Ω resistor
A1/A2	AP/A	R7, R8	2	4			3K
A1/A2	AP/A	R22	1	2			10KΩ resistor
A1/A2	AP/A	R23	1	2			20KΩ resistor

Assy	Sub Assy	Ref Des	Qty per Assy	Total Qty	Manu-facturer	Part Number	Description
A1/A2	AP/A	R24	1	2			6.8KΩ resistor
A1/A2	AP/A	U1-U5	5	10	TI	LM4562NA	Hi performance dual op amp, DIP
A1/A2	AP/A	VR1-VR8		0			(Part of U15 and U16 on Part B)
A1/A2	AP/B	C1, C3, C5	3	6			100uF / 25V electrolytic cap
A1/A2	AP/B	C2, C4, C6	3	6			1uF / 25V tantalum cap
A1/A2	AP/B	J3	1	2			15X2 Header Rcpt, .1" spacing, board mount
A1/A2	AP/B	U16, U17	2	4	Analog Devices	AD8403AN10	4-channel digital potentiometer, 10KΩ, DIP
A1/A2	VTA	C41	1	2			10uF / 16V electrolytic cap
A1/A2	VTA	C42, C43	2	4			10uF / 250V electrolytic cap
A1/A2	VTA	C44, C45	2	4			.1uF / 250V ceramic cap
A1/A2	VTA	C46, C47	2	4			500uF / 50V electrolytic cap
A1/A2	VTA	D1 - D3	3	6	ON Semicon	1N5946BRLG	75V / 3W zener diode
A1/A2	VTA	J2	1	2			1X2 Header Rcpt, .1" spacing, board mount
A1/A2	VTA	J4	1	2		TBD	3 posn keyed rcpt, plastic board mount
A1/A2	VTA	R41	1	2			20KΩ resistor
A1/A2	VTA	R42	1	2			5KΩ resistor
A1/A2	VTA	R43, R44	2	4			33KΩ resistor
A1/A2	VTA	R45, R46	2	4			25KΩ mechanical trimmer potentiometer
A1/A2	VTA	R47	1	2			4KΩ resistor
A1/A2	VTA	R48, R49	2	4			12KΩ resistor
A1/A2	VTA	R50	1	2			30KΩ / 5W power resistor
A1/A2	VTA	R51	1	2			1MΩ resistor
A1/A2	VTA	R52, R57	2	4			100KΩ resistor
A1/A2	VTA	R53, R56	2	4			1KΩ resistor
A1/A2	VTA	R54, R55	2	4			220KΩ resistor
A1/A2	VTA	R59	1	2			1Ω resistor
A1/A2	VTA	R60, R61	2	4			400Ω / 5W power resistor
A1/A2	VTA	T1	1	2	Hammond Mfg	125E	Impedance matching transformer
A1/A2	VTA	V1, V2	2	4	JJ Electronic	6L6GC	Beam power pentode vacuum tube

Assy	Sub Assy	Ref Des	Qty per Assy	Total Qty	Manu-facturer	Part Number	Description
A1/A2	VTA	V3	1	2	Electro-Harmonics	12BH7	Dual triode vacuum tube
A3	PS/M	C1, C2	2	2			1000uF / 25V electrolytic cap
A3	PS/M	D1	1	1			400 V / 1A bridge rectifier
A3	PS/M	J1	1	1			3 posn keyed rcpt, plastic board mount
A3	PS/M	J2	1	1			4X1 Header Rcpt, .1" spacing, board mount
A3	PS/M	J3a, J3b	2	2			15X2 Header Rcpt, .1" spacing, board mount
A3	PS/M	J4	1	1			20X2 Header Rcpt, .1" spacing, board mount
A3	PS/M	R1a, R4a	1	2			117Ω resistor (or closest)
A3	PS/M	R1b, R2b, R3b, R4b, R6b	1	5			1KΩ resistor
A3	PS/M	R2a	1	1			85Ω resistor (or closest)
A3	PS/M	R3a	1	1			220.8Ω resistor (or closest)
A3	PS/M	R5a	1	1			245Ω resistor (or closest)
A3	PS/M	R5b	1	1			400Ω resistor (or closest)
A3	PS/M	R6a	1	1			138Ω resistor (or closest)
A3	PS/M	R7a	1	1			700Ω resistor (or closest)
A3	PS/M	R7b	1	1			5Ω resistor (or closest)
A3	PS/M	U1	1	1		LM7812CT	12V / 1A 3-terminal regulator, TO-220
A3	PS/M	U2	1	1		LM7912CT	-12V / 1A 3-terminal regulator, TO-220
A3	PS/M	U3, U4	2	2		LM337KCT	3-terminal negative adjustable regulator, TO-220
A3	PS/M	U5 - U7	3	3		LM317KCT	3-terminal positive adjustable regulator, TO-220
A3	PS/M	U8	1	1	ST Micro	STM32F303VCT6	ARM Microcontroller
A3	PS/M	U9-U12	4	4	Vishay	SFH6731	4-channel optocoupler, DIP
A4	HVPS	C1, C2	2	2	Cornell-Dublier	LPX101M400A3P3	100uF / 400V electrolytic cap
A4	HVPS	C3	1	1	Nichicon	LGN2H680MELZ30	68uF / 500V electrolytic cap
A4	HVPS	D1	1	1	Vishay	W10G-E4/51	1000V / 2A bridge rectifier
A4	HVPS	D5-D13	9	9	ON Semicon	1N5942B	50V / 3W zener diode
A4	HVPS	J1	1	1			3 posn keyed rcpt, plastic board mount
A4	HVPS	J2	1	1			4 posn keyed rcpt, plastic board mount

Assy	Sub Assy	Ref Des	Qty per Assy	Total Qty	Manu-facturer	Part Number	Description
A4	HVPS	L1	1	1			1.0H / 500mA inductor
A4	HVPS	R1	1	1			10Ω / 10W resistor, flameproof
A4	HVPS	R2	1	1			20Ω / 3W resistor, flameproof
Development			3	3	ST Micro	STM32F3DISCOVERY	Discovery Board
W1	Chassis	J1-J4	4	8			RCA jack, panel mount
W1	Chassis	P1	1	2			4X2 Header Plug, .1" spacing, discrete contacts
W2	Chassis	J5, J6	2	4			RCA jack, panel mount
W2	Chassis	P1	1	2			1X2 Header Plug, .1" spacing, discrete contacts
W3	Chassis		1	2	TBD	TBD	30-conductor ribbon cables, 15x2 plugs
W4	Chassis	P1	1	1	TBD	TBD	
W4	Chassis	P2, P3	2	2			3-posn keyed plug (Mates with A1J4, A2J4)
W4	Chassis	P4	1	1			4-posn keyed plug (Mates with A4J2)
W5		P1, P2	2	2			20X2 Header Plug, .1" spacing, discrete contacts
A5	Chassis		1	1			12V / 3A switching power supply module
F1	Chassis		1	1			Panel-mount fuse holder
J1	Chassis		1	1			IEC320-C14 type panel-mount connector
K1	Chassis		1	1	TBD	TBD	Relay, TBD
P1, P3	Chassis		1	1			3-posn keyed plug (Mates with A3J1, A4J1)
P2	Chassis		1	1			4X1 Header Plug, .1" spacing, discrete contacts
T1	Chassis		1	1	TBD	TBD	120V:30VCT ???A transformer

Table 39

6.4: Software Used

National Instruments Multisim 12.0 Student Version was used for hardware initial circuit design and simulation.

National Instruments Multisim 12.0 Education Version was used for PC layout.

Microsoft Office 2007 and 2010 was used for all documents, graphs and spreadsheets.

Figure 1 was created using Sierra Home Architect, version 4.

Figure 2 was created using Blender, an open-source 3D graphics package.

The demonstration box built for the Senior Design 2 presentation was designed using IMSI TurboCAD, version 12.

Software for the Microcontroller CCA was developed using Atollic True Studio ARM Lite 4.0.1

Software for the digital touch interface used to temporarily replace the touchscreen was developed using TI Code Composer Studio.

The Generation 2 cabinet design was developed using AutoCAD 2011.

6.5: Work Split

Stephen – Audio Processor design, layout, fabrication and test. MCU fabrication. HVPS and Filament Supplies design, fabrication and test.

Jason – MCU design, layout and test. Approximately ½ of application software and test

Raphael – Low Voltage Power Supply design. Approximately ½ of application software and test

Advanced Circuits (http://www.4pcb.com/) fabricated the all PCB (fiberglass and copper only)

Nathan Bodnar (UCF Amateur Radio Club) soldered the STM32F303VCT6 to the MCU CCA.

Parts were procured from the following sources:
Digi-Key: 90%
Found at home or in the lab: 5%
Mouser: 2%
Skycraft: 1%
Texas Instruments: 1% (factory samples of voltage regulators)
Radio Shack: <1% (a few resistors)
eBay: <1% (The LCD panel ordered from a company in Hong Kong)

7: Appendix

7.2.1: JE Audio

JE Audio
Hi Stephen, You are now given the permission from JE Audio to use the VM60 photo in your Sr Design report. You mentioned the project is about tube amp. I

wonder what kind of design and product that you are going to do or have developed. You do not need to send
Fri 3/22
Friday, March 22, 2013 1:29 AMJE Audio [info@je-audio.com]
To:
nicholsse@knights.ucf.edu

You replied on 3/23/2013 1:40 PM.
Hi Stephen,
You are now given the permission from JE Audio to use the VM60 photo in your Sr Design report.
You mentioned the project is about tube amp. I wonder what kind of design and product that you are going to do or have developed. You do not need to send us the report. If you can, just say few words about any new or unique features about the tube amp. And why is there a need to use the VM60 photo?
If I do not hear from you, wish you best in your project.
Regards,
John Lam
JE AUDIO

7.2.2: JJ Electronic

Stephen,

thank you for choosing JJ Electronic!

I give consent to the use of data from our web site.

Best regards,
Julia Jurcova
sales
JJ Electronic
A. HLinku 4
ZIP: 022 01
Cadca, Slovak Republic
tel.: +421 41 4304 120
fax.: +421 41 4335 370
info@jj-electronic.com
www.jj-electronic.com

7.2.3: Z. Vex Effects

Hi Stephen,

If you need a higher resolution imag= of the iMP AMP, please don't hesitate to ask, I will send one your way :)=br>
Hannah

On Thu, Mar 21, 2013 at 2:27 PM, Zack Vex <zack@zvex.com> wrote:=br>
Hi Stephen,

Of course you have my permission. I consider it an honor. I hop= you
get an A. 8^D

Rock on,
Zack

7.2.4: STMicroelectronics

Tuesday April 16, 2013

Jason,
Please feel free to use any information you obtain from st.com for your project and its documentation.

Good luck on your senior project.

Richard Steele
Product Marketing Engineer, South East territory
STMicroelectronics
30 Corporate Drive
Suite 300
Burlington, MA 01803

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